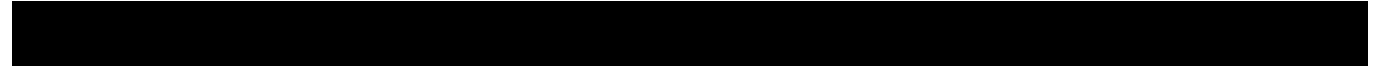




**MOTOROLA**

# GPS Simulcast Installation Manual

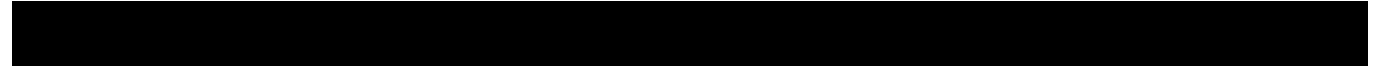






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# GPS Simulcast Installation Manual



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This warranty extends only to individual products: batteries are excluded, but carry their own separate limited warranty.



# Contents

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## **Chapter 1 Overview 1-1**

What Is GPS Simulcast? 1-5

Simulcast System Configurations 1-7

Analog Mode Using DSM-IIIs 1-8

Digital Mode 1-21

Mixed Mode Using ASTRO-TAC 1-33

Mixed Mode Using ASTRO-TAC 3000 1-47

## **Chapter 2 Rack Installation 2-1**

Unpacking And Inspection 2-2

Rack Installation 2-3

## **Chapter 3 Conventional Simulcast Controller Interface 3-1**

Description 3-2

Theory of Operation 3-3

CSCI User Interface Switches, Controls, And Connectors 3-8

Installing The CSCI 3-15

Troubleshooting 3-19

CSCI  
(QTN6066B) Parts List 3-29

## **Chapter 4 Simulcast Distribution Amplifier 4-1**

Description 4-2

Theory of Operation 4-4

Troubleshooting 4-7

Switch Settings Reference Guide 4-8

SDA Edge Connector Pinout 4-9

SDA Parts List/Schematic Diagram 4-11

Simulcast Distribution Amplifier (QRN4878A) Parts List 4-15

## **Chapter 5 DSM-II Module 5-1**

Description 5-2

DSM-II Specifications 5-5

Description Of DSM-II Network Configurations 5-6

DSM-II Card Programming Instructions 5-18

DSM-II Module Installation 5-29

DSM-II Configuration Programming Instructions 5-31

DSM-II Troubleshooting Instructions From the DSM-II Test Window 5-50

DSM-II Phasing Using 1 PPS (GPS Operation) 5-63

DSM-II Pinouts 5-67

## **Chapter 6 Data Broadcast Box 6-1**

Description 6-2

Installation 6-4

Troubleshooting 6-5

## **Chapter 7 Optimization 7-1**

Why Simulcast Optimization is Required 7-2

Optimization Requirements 7-8

Analog System Optimization 7-9

Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000) 7-57

Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping) 7-71

Digital (ASTRO) System Outbound Optimization 7-75

Mixed System Optimization 7-82

Performance Verification 7-82

Optimization Using A Remote Console 7-83

# Overview

## About This Manual

This manual describes the configuration, operation, and optimization of equipment unique to analog simulcast systems using Digital Simulcast Modems, second generation (DSM-IIs) (digital transport; Trunked and Conventional systems), ASTRO<sup>®</sup> simulcast systems (Trunked and Conventional systems), and mixed mode simulcast systems (Trunked systems only). In addition, the manual contains sections on the DSM-II, Conventional Simulcast Controller Interface (CSCI), Data Broadcast Box (DBB), and the Simulcast Distribution Amplifier (SDA).

For mixed mode systems that use either analog microwave (dual path) or Siemens channel banks (digital path), refer to *Trunked Radio System Dual Path and Digital Path Simulcast Equipment* (see Recommended Reference Publications, page 1-2), for the configuration, operation, and optimization for the analog portion of a mixed mode channel. This manual also contains a section on the Universal Simulcast Controller Interface.

An overview of this manual is provided in Table 1-1.

**Table 1-1 Manual Contents and Overview**

Chapter	Description
Overview	Describes the scope of this manual and provides an overview of the equipment and configurations covered.
Rack Installation	Provides general instructions for setting up equipment racks.
Conventional Simulcast Controller Interface (CSCI)	Describes the CSCI, how to install it, and provides troubleshooting information.
Simulcast Distribution Amplifier (SDA)	Describes the SDA and how to install it, and provides troubleshooting information.
DSM-II Module	Describes DSM-II (Digital Simulcast Modem) and how to install it, and provides procedures for field troubleshooting.
Data Broadcast Box (DBB)	Describes the DBB and how to install it, and provides procedures for field troubleshooting.
Optimization	Describes optimization and provides optimization procedures for analog, digital, and mixed systems.

## Recommended Reference Publications

- ❑ *QUANTAR and QUANTRO Service Manual*  
Motorola manual 68P81095E05
- ❑ *QUANTAR and QUANTRO Station Radio Service Software Guide*  
Motorola manual 68P81085E35
- ❑ *TeNSr Manual*  
Motorola manual 68P80309F69
- ❑ *FNE Installations R56* (Quality Standards)  
Motorola manual 68P81089E50
- ❑ *Trunked Radio System Dual Path and Digital Path Simulcast Equipment*  
Motorola manual 68P81081E60

This manual includes the following documents with their own respective manual numbers. Only those documents referenced in the GPS Simulcast Installation Manual are listed separately below.

- SCI Power Supply TPN1153A/TPN1170A (68P81039E36) (1 page schematic drawing)
- Dual Path Simulcast Optimization (68P81081E71)
- Dual Path Simulcast System (68P81126E83)

## Terms Used in This Manual

Acronyms and other terminology used in this manual are defined in Table 1-2.

Table 1-2 **Acronyms and Particular Terms used with Simulcast System**

Acronym/Term	Definition
ADPCM	Adaptive Differential Pulse Coded Modulation. A server resource card used in the TeNSr channel banks also for ADPCM voice compression of a 64 kbps time slot and ASTRO audio combining them into a DS0 at the output.
AEB	Ambassador Electronics Bank. Motorola product name for the electronic core of the audio switch in a trunked system that routes/combines audio sources.
ASTRO	Motorola tradename for proprietary digital simulcast system.
CSCI	Conventional Simulcast Controller Interface
CVSD	Continuously Variable Slope Data. A sophisticated method of analog-to-digital conversion to encrypt signals for secure operations.
DBB	Data Broadcast Box. A line sharing or multiplexing device that receives and passes asynchronous signals between the zone controller and the digital interface units.
DIU	Digital Interface Unit. Device used in ASTRO or mixed-mode ASTRO systems for encryption/decryption of audio to or from the console.
DPL	Digital Private Line. A continuous subaudible digital code used in conventional systems for squelch.
DSM	Digital Simulcast Modem. DSM is used with Siemens channel bank.



Table 1-2 **Acronyms and Particular Terms used with Simulcast System (continued)**

Acronym/Term	Definition
DSM-II	Digital Simulcast Modem, second generation. DSM-II has a patented feature that automatically adjusts and maintains the precise time delay required by the simulcast system. The DSM-II makes use of signals from the GPS system to provide this feature. DSM-II is used with TeNSr channel bank.
DSU	Data Service Unit. A modem used in a Siemens channel bank to transport V.24.
DVP	Digital Voice Protection. An encryption method used to prevent unauthorized listeners from hearing certain voice transmissions.
ESDS	Electrostatic Discharge Sensitive
Failsoft	The failure mode of a trunked radio system, in which each channel acts as a conventional channel. A trunked system enters Failsoft to allow users to continue to communicate if the trunking central controller fails.
FRED	Four-level Recovery Encode/Decode. A modulation scheme used for transmitting SECURENET <sup>®</sup> signals in simulcast system. The use of this modulation scheme allows for increased site separation.
FSK	Frequency Shift Keying. Frequency modulation in which transmitted data symbols (bauds) are represented by shifts to predetermined frequencies above and below carrier.
GPS	Global Positioning System. A network of satellites in a geosynchronous orbit, whose precise signals are used to provide timing synchronization for the simulcast equipment described in this manual.
LC	Link Control. A data field used for low bit-rate control signaling over an ASTRO channel. LCs provide information that the subscriber uses to stay on the voice channel during a call, including the digital equivalent of analog lowspeed data. In addition, the LC field also contains PTT-ID information.
PCM	Pulse Code Modulation. Modulated sequence of pulses representing a quantized input. High-end applications of PCM include transferring GPS information to ground computers.
PL	Private Line. A continuous subaudible tone used in conventional systems for squelch. This is referred to as Continuous Tone-Controlled Squelch System (CTCSS).
PON	Prime Optimization Node. A computer system used to optimize and maintain simulcast audio distribution.
PPS	Pulse per second
RDM	Remote Delay Module. A module used in simulcast systems and used to optimize the simulcast audio.
RF Capture	An FM phenomenon in which a radio receives multiple signals, with one of the signals having significantly greater signal strength than the others. The strongest signal "captures" the FM receiver. This results in the recovered audio being solely due to this signal, rather than the sum of all the received signals.
RF Non-Capture	A situation where the radio receives multiple signals, with none of the signals being strong enough to produce RF capture. In this case, the recovered audio is a result of summing all the received signals.
RSS	Radio Service Software
SCI	Simulcast Controller Interface

Table 1-2 **Acronyms and Particular Terms used with Simulcast System (continued)**

Acronym/Term	Definition
SDA	Simulcast Distribution Amplifier
SECURENET®	An analog signal converted to digital and then encrypted with CVSD encryption in Motorola secure systems.
Simulcast	A wide-area trunked system configuration that uses multiple transmitter and receiver sites to extend coverage of the system. All the corresponding channel numbers at all the sites use the same frequency.
SRU	Sub-Rate Unit. A module used in a TeNSr channel bank to transport V.24.
SSA	Simulcast Serial Adapter
SSB	Single-Sideband
T1/E1	Standard telephone transmission line/time-division multiplexing scheme. (T1 is USA; E1 is European)
TDATA	Transmitter Data. Control line pair from the central controller supplying the trunking data to the repeater.
TDM	Time Division Multiplex
TeNSr	Telecommunications Network Server. Multi-functional access device used to combine the capabilities of a channel bank and signal converter.
TPTT	Trunking Push-to-Talk
USCI	Universal Simulcast Controller Interface. An interface that is part of the simulcast audio distribution network.

## Service Information

Contact the Motorola System Support Center for the following:

- ☐ Repair or warranty information because you experienced trouble with the equipment described in this manual.
- ☐ Assistance in sending the boards back for repair. (None of the GPS Simulcast circuit boards are field repairable.)

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## What Is GPS Simulcast?

Global Positioning System (GPS) simulcast uses signals from the Navstar GPS system to synchronize the audio from multiple transmitters. The GPS system consists of 24 satellites orbiting the earth which send precise time reference signals to GPS receivers on earth. The GPS receivers use the satellite signal to derive a high-precision 1 pulse-per-second (PPS) signal which is used in the simulcast system for time launching. By *launching* signals at exactly the same time from multiple sites, destructive interference of the transmitted signals in overlap areas is minimized. In addition to controlling the launch, the use of GPS allows for variance in delay of the distribution network (T1/E1).

In a digital system, the comparator/base station controls the synchronization of the signals to be transmitted. In an analog system, the prime and remote DSM-II ports control the synchronization of the signals to be transmitted.

### Analog Component (DSM-II) Time Stamp and Delay Compensation

The GPS 1 PPS signal is used to measure the path delays for the Telecommunications Network Server (TeNSr) equipment that is used to transport the analog signals. At each site (prime and remote), a 1 PPS signal will be generated by the site time/frequency reference. The 1 PPS signals at each site are within 1  $\mu$ sec of each other (maximum). When the prime site receives its 1 PPS signal, the DSM-II will time-stamp the outbound path. When the remote site DSM-II receives this time stamp, it compares it to the current time, as determined from its local 1 PPS signal. This time difference is subtracted from the programmed DSM-II launchtime offset delay parameter. This end result is the length of time the DSM-II will hold the received audio in a buffer before putting it on the voice channel. The amount of buffer time represents what remains of the required delay once the audio reaches the remote site DSM-II.

The site standard GPS receiver provides a 1 PPS signal to the DSM-II, located within the TeNSr channel bank. Within the system, a DSM-II is used at the prime site and the remote site. Therefore, a GPS receiver is required at the prime and remote sites to provide a 1 PPS signal to the DSM-II cards.

### ASTRO Digital Component Time Stamp and Delay Compensation

The ASTRO-TAC<sup>®</sup>/ASTRO-TAC 3000<sup>™</sup> comparator and ASTRO base station use the 1 PPS signal in a slightly different manner than the DSM-II. The GPS receiver provides the 1 PPS signal. An ASTRO signal is first time-stamped at the comparator with a number representing the number of ticks past the current 1 PPS mark seen by both the station and comparator. This information is embedded with the ASTRO voice/data and sent to the simulcast station. The comparator calculates the time-stamp value by adding the comparator launchtime offset RSS parameter to the current time at the comparator. The comparator launchtime offset value is the sum of the station processing time and the transmission path

**What Is GPS Simulcast?**

delay. The simulcast station receives the ASTRO voice/data with the embedded time stamp and places the information into a buffer where it is held until the specified time arrives. At this time, the ASTRO digital information is launched onto the RF channel.

If a station offset adjustment has been entered into station Radio Service Software (RSS) to fine-tune the coverage delays, the simulcast station adds this value to the time stamp received from the comparator. This can be used to delay the signal from a site so that reception in the equal signal, non-capture area will consist of both signals with the same modulation phase, thus minimizing bit errors. When there is no capture, simulcast distortion will exist if the relative delay between signals is greater than approximately 100  $\mu$ sec for an ASTRO wide-pulse signal and 50  $\mu$ sec for an ASTRO narrow-pulse signal.

**NOTE**

These values change depending on system topology and coverage reliability level. The values are custom-tailored by the Area Systems Engineer (ASE).

The station offset adjustment adds fine adjustments (steps of 100 nsec) to the launchtime. The comparator launchtime has steps of 1 msec. Again, the only purpose of the comparator launchtime adjustment parameter is to tell the station how long to hold the digital information prior to launch. When the digital voice signals arrive at the simulcast stations, they are held until the launchtime specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel. If the station misses the specified launchtime (due to an unexpected infrastructure delay or improper system configuration), an error is written to the station error log. In SmartZone<sup>®</sup> systems, a technician diagnostic message will also be generated at a zone manager terminal.

**Practical Considerations In Simulcast Digital Loop Switch Operation**

In a simulcast system that uses digital loop switching, the launchtime (ASTRO systems) or MAX Delay (analog systems with DSM-IIIs) must be set so that it is higher than the delay for the longest path in the microwave loop system. This ensures that no matter which route a signal takes to any given transmitter, it will arrive before all the sites have launched. The same principle applies for non-loop, alternate routing distribution methods.

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## Simulcast System Configurations

There are four different GPS simulcast system configurations. These are:

- ☐ Analog Mode Using DSM-IIIs
- ☐ Digital Mode
- ☐ Mixed Mode Using ASTRO-TAC
- ☐ Mixed Mode Using ASTRO-TAC 3000

These configurations are discussed respectively in detail in later sections of this chapter.

Systems that use GPS can use either dedicated T1/E1 lines, or T1/E1 lines that use alternative routing for the inter-site communications.

<b>NOTE</b>
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If a mixed mode simulcast system uses either analog microwave or Siemens channel banks, then the GPS system performs the phasing for the ASTRO portion of the system only. The phasing for the analog portion in this case is accomplished using Prime Optimization Node (PON).

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## Analog Mode Using DSM-IIs

For analog simulcast systems that use GPS, dedicated T1/E1 lines or T1/E1 lines with alternative routing can be used to distribute the audio and trunking data (Trunking systems) or PL/DPL (Conventional systems) to the remote sites. Since a digital medium is being used, this is a digital path system where the trunking data or PL/DPL is summed with the audio at the prime site. Second generation Digital Simulcast Modems (DSM-II) transfer the data to the remote site. Figures 1-1 through 1-6 show analog systems in various configurations.

For each remote site channel, one DSM-II port is used at the prime site (Tx DSM-II) and one at the remote site (Rx DSM-II). (One DSM-II card contains four full duplex ports.) The DSM-II card resides in the TeNSr channel bank. The DSM-II is configured through programming in one of two ways:

- ☐ Non-broadcast Mode. Non-broadcast mode supports clear only channels.
- ☐ Broadcast Mode. Broadcast mode supports clear, SECURENET 2L, and SECURENET Four-level Recovery Encode/Decode (FRED) channels.

In a clear only system, the DSM-II can be configured for either the non-broadcast mode or the broadcast mode. For small systems using only one or two channel banks at the prime site, the broadcast mode is easy to configure. However, as a system gets larger with more channels and more sites, channel bank programming for the broadcast mode becomes more complicated. In this situation, a system designer may choose to use the non-broadcast mode. In general, there is no set formula for using either the non-broadcast mode or the broadcast mode. Chapter 5, “DSM-II Module,” provides detailed information about the use and setup of these various modes.

### Non-broadcast Mode

Figures 1-1 on page -15 and 1-2 on page -16 show trunked and conventional non-broadcast mode analog subsystems, respectively. The following paragraphs describe a non-broadcast mode subsystem.

#### Transmit Path

The transmit audio path begins with the comparator and ends at the remote sites with the base stations/antennas. The comparator can accept multiple receiver inputs (inbound path of the system) or an input from the console. The comparator selects (votes) from the receiver inputs and routes the selected input to the comparator transmit audio output. The comparator also routes the console input to the transmit audio output. The comparator can be programmed to grant priority to the console or to the receiver. When activity is present at the receiver inputs and at the console, the comparator grants priority to the input programmed for priority. This input is then routed to the transmit audio output. The input not programmed for priority is not routed to the transmit audio output. The comparator transmit audio output is connected to the input of the Universal

Simulcast Controller Interface (USCI) or Conventional Simulcast Controller Interface (CSCI).

The USCI (or CSCI) pre-emphasizes and limits the transmit audio signal. In a 900 MHz simulcast system, the USCI (or CSCI) compresses the transmit audio signal. In a trunking system, once the transmit audio signal has been processed, it is summed with the lowspeed data or failsoft word and routed through a low pass filter. In a conventional simulcast system, the CSCI sums the Private Line (PL) / Digital Private Line (DPL) with the audio and then passes it through the lowpass filter. From the lowpass filter output, the transmit audio is routed to the USCI (or CSCI) output.

When the channel is a control channel (trunking system only), the transmit audio path is muted. The control channel data is passed through the lowpass filter, followed by the USCI (or CSCI) output.

In a conventional system, there is no control channel. Therefore, the audio path will not be muted in a conventional system.

The output of the USCI (or CSCI) is connected to the Simulcast Distribution Amplifier (SDA). The SDA is a unity gain device that accepts one 600 $\Omega$  input and produces ten, 600 $\Omega$  outputs.

<b>NOTE</b>
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In a conventional system, there can be more than 10 sites. For conventional systems with more than 10 sites, the SDAs are cascaded.

The output from the SDA is connected to the Tx DSM-II port. Each Tx DSM-II port at the prime site is programmed as a **prime**. For a co-located site, the Tx DSM-II port is programmed as a **loopback**. (See Chapter 5, "DSM-II Module," for details.)

<b>NOTE</b>
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The loopback port is also the Rx DSM-II port, since the port is internally looped on itself.

The Tx DSM-II port converts the transmit audio voltage to digital samples to be placed in the frame structure of the DSM-II. The DSM-II frame structure is placed in a time slot of a T1/E1 signal through Time Division Multiplexing (TDM). The T1/E1 signal can be applied to the input of a digital microwave or to a leased T1 line.

At the remote site, the received T1/E1 signal is demultiplexed into the individual time slots and applied to the corresponding DSM-II port at the remote site. The Rx DSM-II port converts the digital samples back to the analog voltages to produce the desired audio frequencies. The Rx DSM-II at a remote site is programmed as a

## Analog Mode Using DSM-IIs

**rem.** At the remote site, the Rx DSM-II adjusts the amplitude and phase of the transmit audio. The transmit audio and data are then routed to the transmitter.

The push-to-talk (PTT) signal for an analog simulcast system using DSM-IIs originates in the prime site central controller during trunking operations and is sent to the USCI. In a conventional system, the CSCI will accept a PTT indication from the comparator (Vote Indicate). From the CSCI/USCI, the PTT is routed to the M1-Lead input of the SDA. The SDA produces 10 M1 outputs that are connected to the Tx DSM-II port. When the Rx DSM-II receives the M1 from the Tx DSM-II, it produces an E1-Lead output. The E1-Lead output of the Rx DSM-II port is connected to the external PTT line of the base station.

### Receive Path

The receive path of the system begins at the base station, which receives the inbound RF signal from mobiles and portables. The base station demodulates the received signals and routes them to the wireline output connected to the transmit side of the DSM-II port (Tx DSM-II port for the inbound path) at the remote site for transmission back to the prime site.

At the prime site, the output of the Rx DSM-II port is routed to the comparator. There is no time stamping and amplitude optimization adjustment on the inbound path.

### Failsoft Operation

During failsoft, the PTT originates at the USCI and is routed from the USCI to the M1-Lead input of the SDA. The USCI generates the failsoft data and failsoft alert tone and sums them together with the transmit audio signal for routing to the SDA.

## Broadcast Mode

Figures 1-3 and 1-4 show trunked and conventional broadcast mode analog subsystems, respectively. Figures 1-5 and 1-6 show a trunked and conventional broadcast mode analog SECURENET 12 kbit system. The following paragraphs describe broadcast mode subsystems. In the broadcast mode, the function of the SDA is incorporated into the DSM-II.

### Transmit Audio

The transmit audio path begins with the comparator and ends at the remote sites with the base stations/antennas. The comparator can accept multiple receiver inputs (inbound path of the system) or an input from the console. The comparator selects (votes) from the receiver inputs and routes the selected input to the comparator transmit audio output. The comparator also routes the console input to the transmit audio output. The comparator can be programmed to grant priority to the console or to the receiver. When activity is present at the receiver inputs and at the console, the comparator grants priority to the input



programmed for priority. This input is then routed to the transmit audio output. The input not programmed for priority is not routed to the transmit audio output. The comparator transmit audio output is connected to the input of the Universal Simulcast Controller Interface (USCI) or Conventional Simulcast Controller Interface (CSCI).

The USCI (or CSCI) pre-emphasizes and limits the transmit audio signal. In a 900 MHz simulcast system, the USCI (or CSCI) compresses the transmit audio signal. In a trunking system, once the transmit audio signal has been processed, it is summed with the lowspeed data or failsoft word and passed through a lowpass filter. In a conventional simulcast system, the CSCI sums PL or Digital Private Line (DPL) with the audio and then passes it through the lowpass filter. From the lowpass filter output, the transmit audio is routed to the USCI (or CSCI) output.

During the SECURENET mode, the 12 kbit data bypasses the pre-emphasis and limiting circuitry. The TDATA from the prime central controller or the PL/DPL is muted. The USCI (or CSCI) accepts the 12 kbit data and routes it through the lowpass filter. The filtered transmit audio is routed to the USCI (or CSCI) output.

<b>NOTE</b>
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900 MHz does not support 12 kbit encrypted voice.

In the case of trunking control channels, the transmit audio path is muted. The control channel data is passed through the lowpass filter followed by the USCI output. (In a conventional system, the audio path is not muted because there is no control channel.)

As stated previously, the function of the SDA is incorporated into the DSM-II when the DSM-II is configured for the broadcast mode. The transmit audio signal is fed into one DSM-II port. This port is programmed as a prime. The prime port (Tx DSM-II) communicates with one remote site DSM-II port (Rx DSM-II). For each additional site, a slave DSM-II port is required at the prime site. The slave port (Tx DSM-II) copies (broadcasts) the outbound 64 kbit stream from the prime port into its outbound 64 kbit stream. The slave port communicates with a DSM-II port (Rx DSM-II) at another remote site.

A slave-loopback port is required for a co-located site. The slave-loopback port get its transmit audio from the prime port in the same manner as the slave port.

The DSM-II converts the transmit audio voltage to digital samples to be placed in the frame structure of the DSM-II. In the SECURENET mode, the DSM-II places the 12 kbit data, as well as the phase of the 12 kbit clock, in the frame structure of the DSM-II. The DSM-II frame structure is then placed in a time slot of the T1/E1 through TDM. The T1/E1 signal can be applied to the input of a digital microwave or a leased T1/E1 line. At the remote site, the received T1/E1 signal is demultiplexed into the individual time slots and applied to the corresponding DSM-II port.

The remote DSM-II port accomplishes the delay and amplitude to be set for the channel for both the clear and SECURENET modes of operation. In the

**Analog Mode Using DSM-IIs**

SECURENET mode, the DSM-II converts the 12 kbit 2L signal to a 12 kbit 4L signal (FRED) for a 4L system. This is described in detail in Chapter 5 – DSM-II Module. At the remote site, the Rx DSM-II port routes the transmit audio and data to the base station.

The PTT signal for an analog simulcast system using DSM-IIs originates in the prime site central controller during trunking operations and is sent to the USCI. During failsoft, the PTT originates at the USCI. In a conventional system, the CSCI will accept a PTT from the comparator (Vote Indicate). From the USCI in either trunking or failsoft conditions or the CSCI (conventional system), the PTT is routed to the M1-Lead input of the prime port (Tx DSM-II). In the broadcast mode, the M-Lead signaling will be broadcasted in the same manner as the audio. The E1-Lead output of the Rx DSM-II port is connected to the external PTT line of the base station.

During a secure call, the comparator provides a code indicate to the USCI (or CSCI). When the USCI (or CSCI) receives the code indicate, it mutes the clear audio path and passes the 12 kbit data, along with the code indicate, directly through to the Prime DSM-II port. The code indicate is required to place the DSM-II in the coded mode of operation. The DSM-II input for the code indicate is the M2 signaling lead. At the remote site, the code indicate signal is not required for the base station.

**Receive Path**

The receive path of the system begins at the base station, where the inbound RF signal from mobiles and portables is received. The base station demodulates the received signal and routes it to the wireline output, which is connected to the transmit side of the DSM-II port (Tx DSM-II) at the remote site for transmission back to the prime site.

**NOTE**

During the SECURENET mode, the remote site DSM-II Tx port requires a code indicate from the base station to place it in the SECURENET mode of operation.

The Rx DSM-II port converts the digital samples back to the analog voltages to produce the desired audio frequencies. In the SECURENET mode, the 12 kbit data is clocked out of the DSM-II according to the phase of the 12 kbit clock from the remote site DSM-II port. The output of the Rx DSM-II port is routed to the comparator.

## SECURENET Equipment

A SECURENET daughterboard is required for the DSM-II at both the prime and remote sites. Through programming of the DSM-II card, the DSM-II can support either 2L simulcast or 4L (FRED) simulcast. A SECURENET simulcast system will only be configured for 2L when the subscribers will not support FRED.

### NOTE

For SECURENET simulcast operations, the DSM-II must be configured for the broadcast mode. The non-broadcast mode does not support SECURENET simulcast operation.

## Failsoft Operation

During analog failsoft in a trunking system, once the transmit audio signal has been processed, it is summed with the failsoft word and passed through a lowpass filter. From the lowpass filter output, the transmit audio is routed to the USCI output.

The PTT signal normally originates in the prime site central controller during trunking operations and is sent to the USCI. However, during failsoft, the PTT originates at the USCI. From the USCI in failsoft conditions, the PTT is routed to the M1-Lead input of the prime port (Tx DSM-II). In the broadcast mode, the M-Lead signaling will be broadcast in the same manner as the audio.

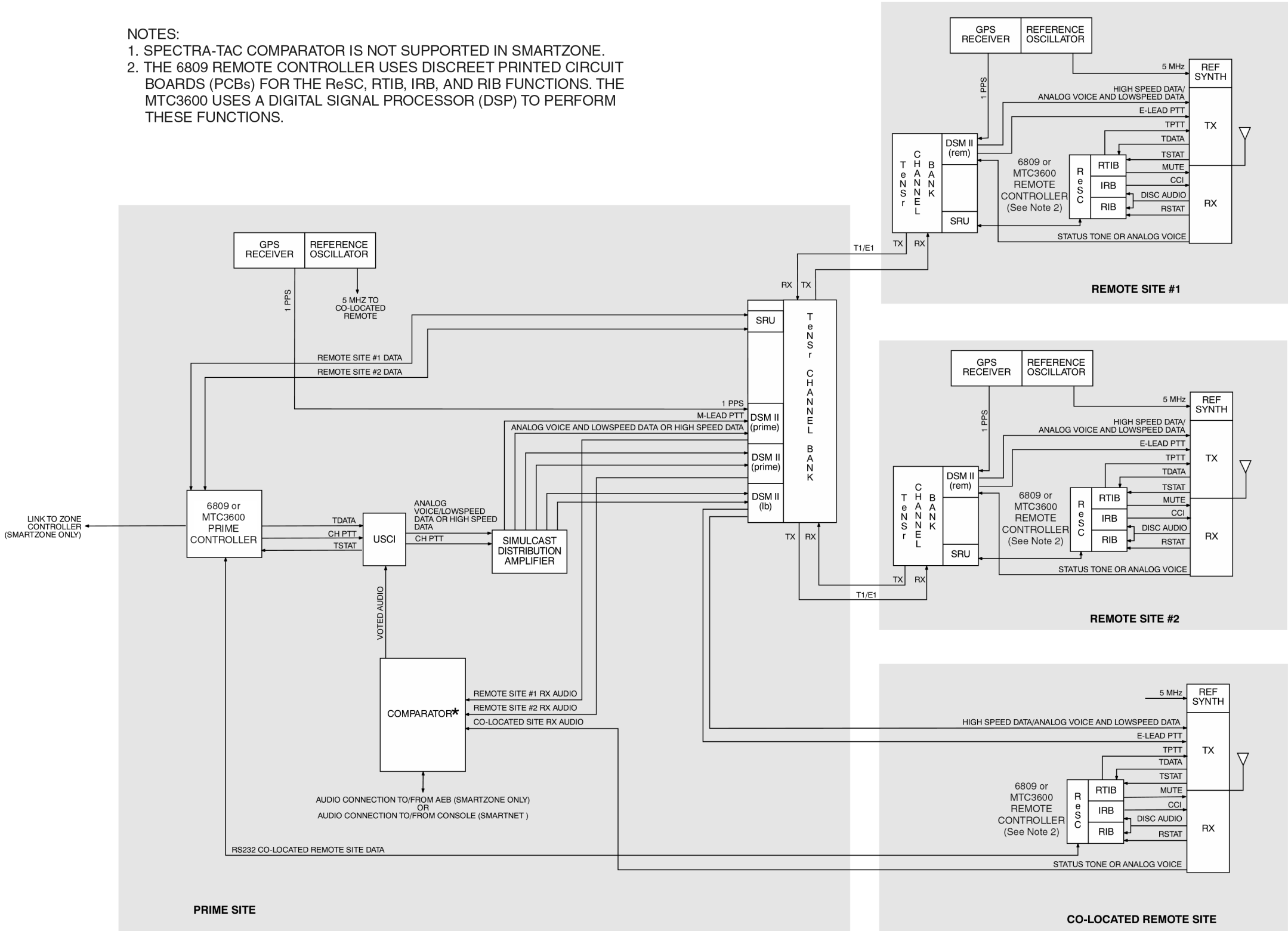
## Extended Broadcast Mode

The extended broadcast mode functions the same as the broadcast mode, but is larger in scale. The broadcasting is done across channel banks.

**Analog Mode Using DSM-IIIs**

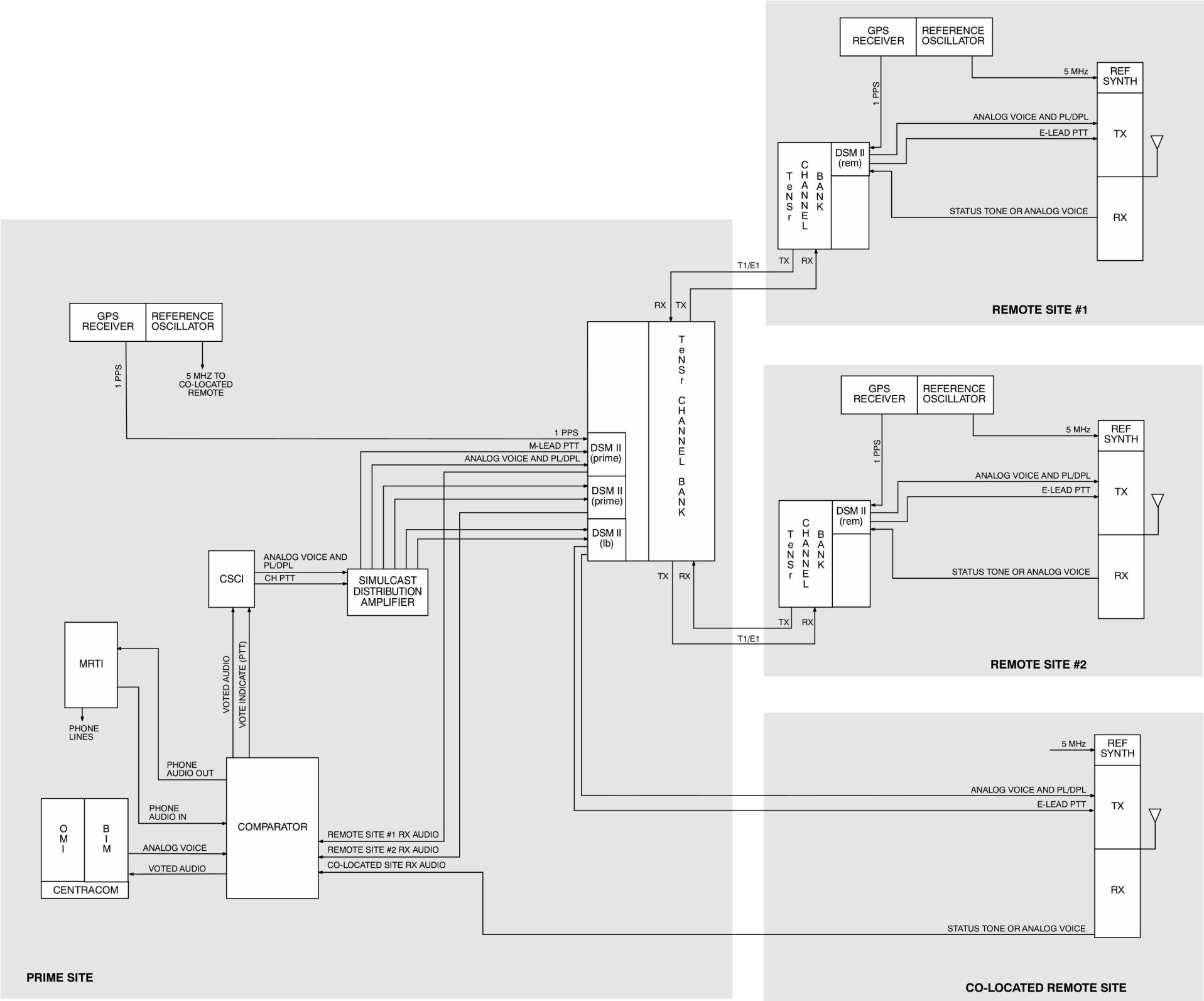
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- NOTES:
- 1. SPECTRA-TAC COMPARATOR IS NOT SUPPORTED IN SMARTZONE.
  - 2. THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.



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082900SKS

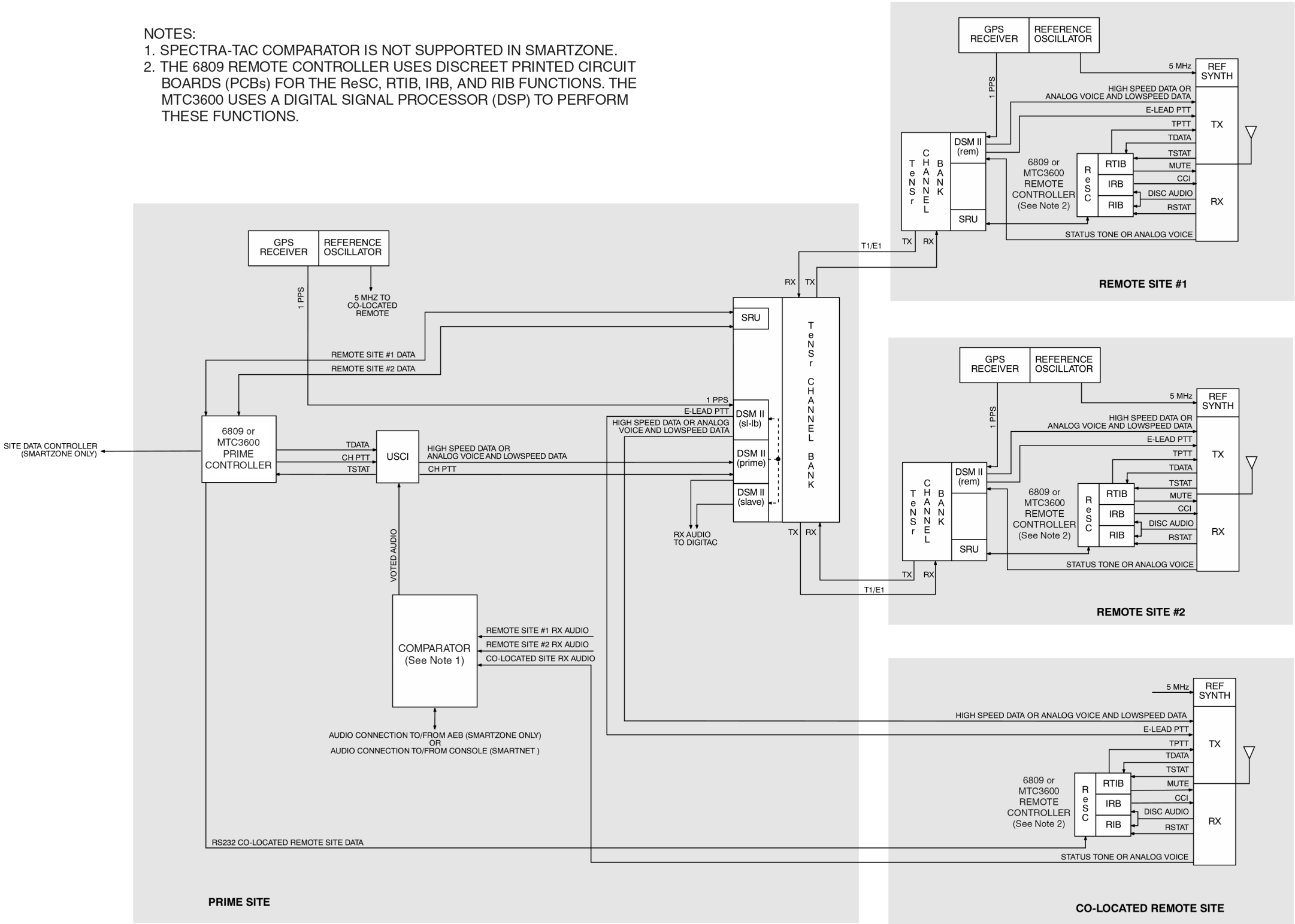
Figure 1-1 Trunked Digital Path Analog Clear Voice (Non-Broadcast) With TeNSr Channel Banks



SIMU046  
102496JNM

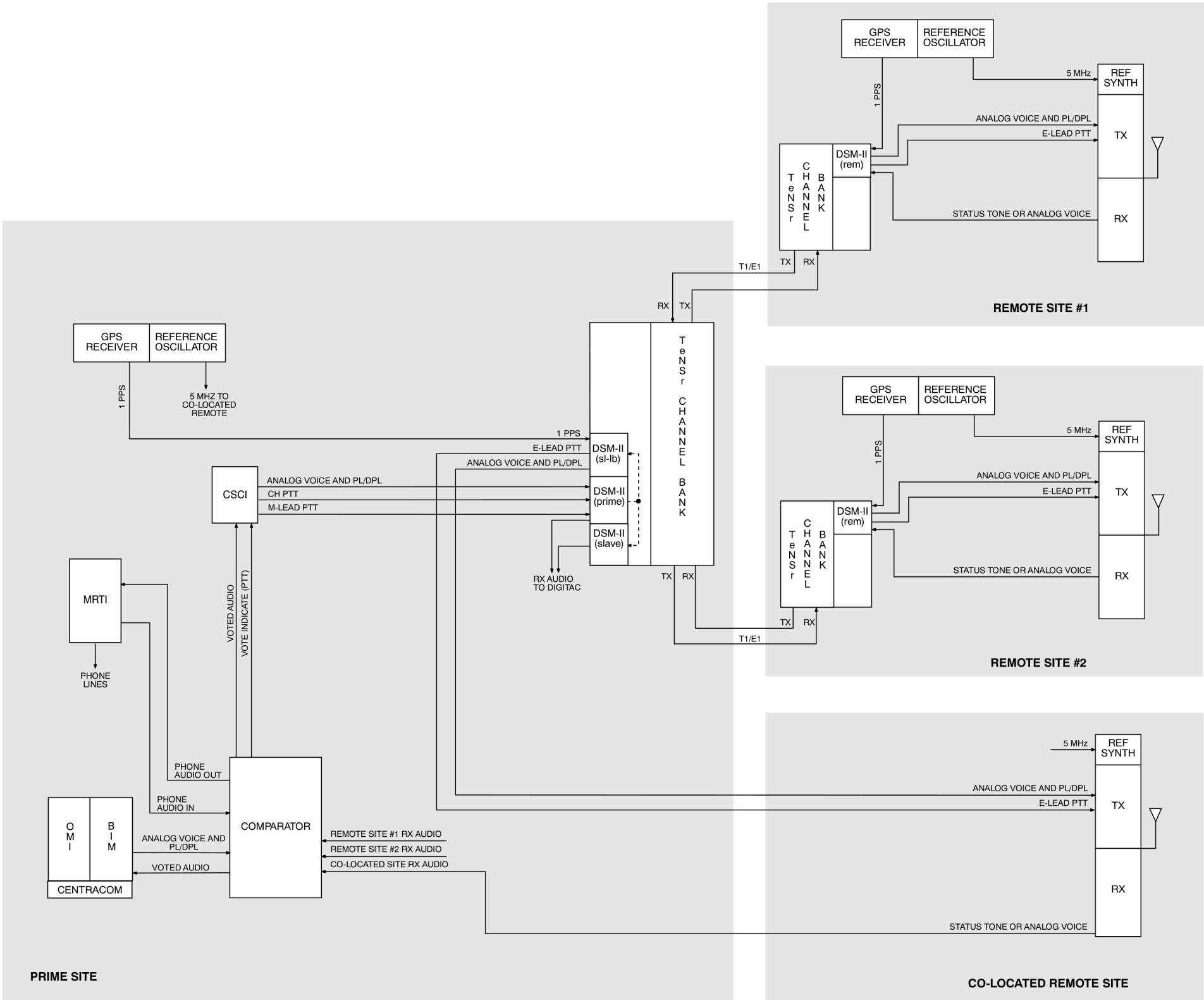
Figure 1-2 Conventional Digital Path Analog Clear Voice (Non-Broadcast) With TeNSr Channel Banks

- NOTES:
- 1. SPECTRA-TAC COMPARATOR IS NOT SUPPORTED IN SMARTZONE.
  - 2. THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.



SIMU003A  
082900SKS

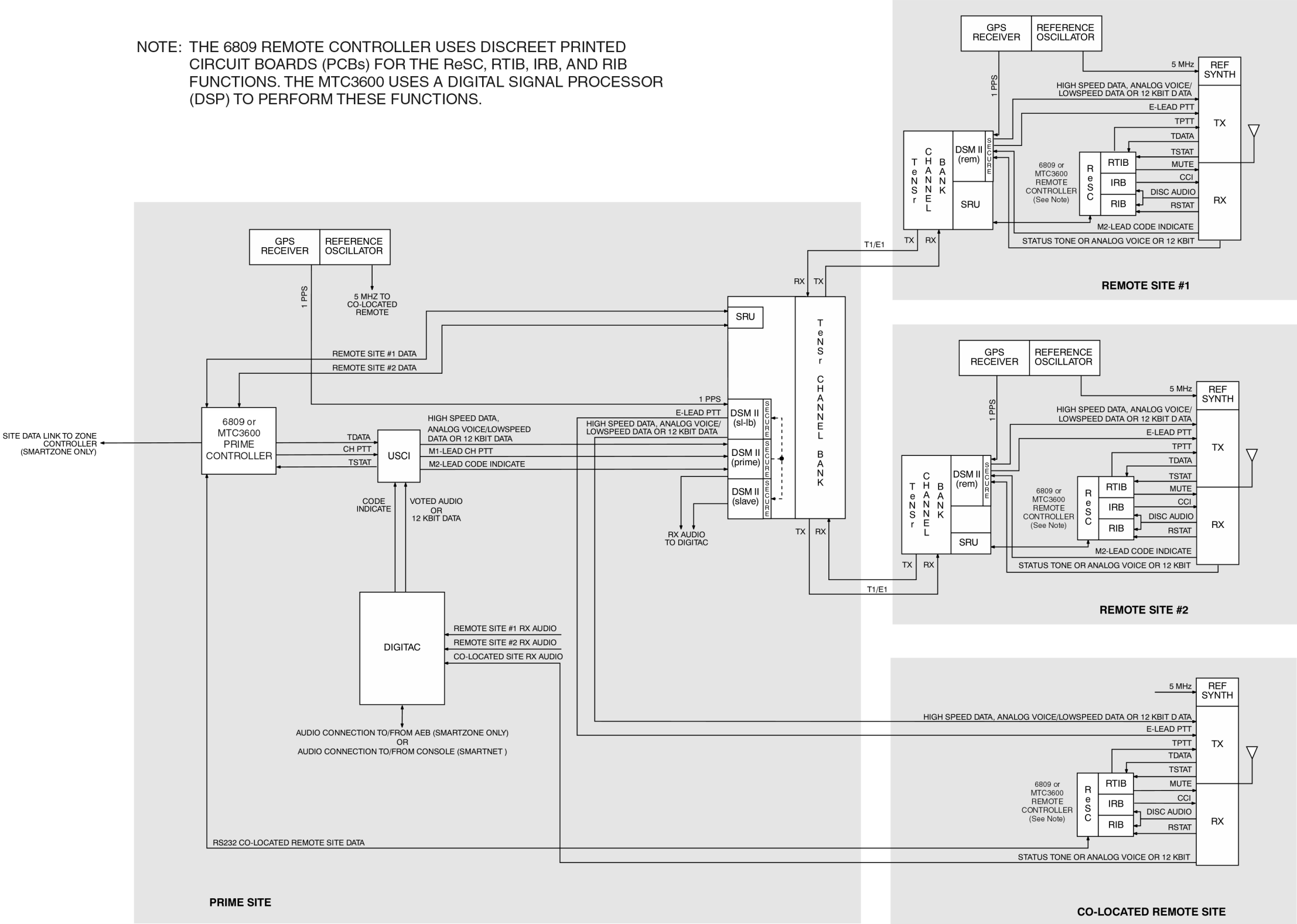
Figure 1-3 Trunked Digital Path Analog Clear Voice (Broadcast) With TeNSr Channel Banks



SIMU047  
110696SN

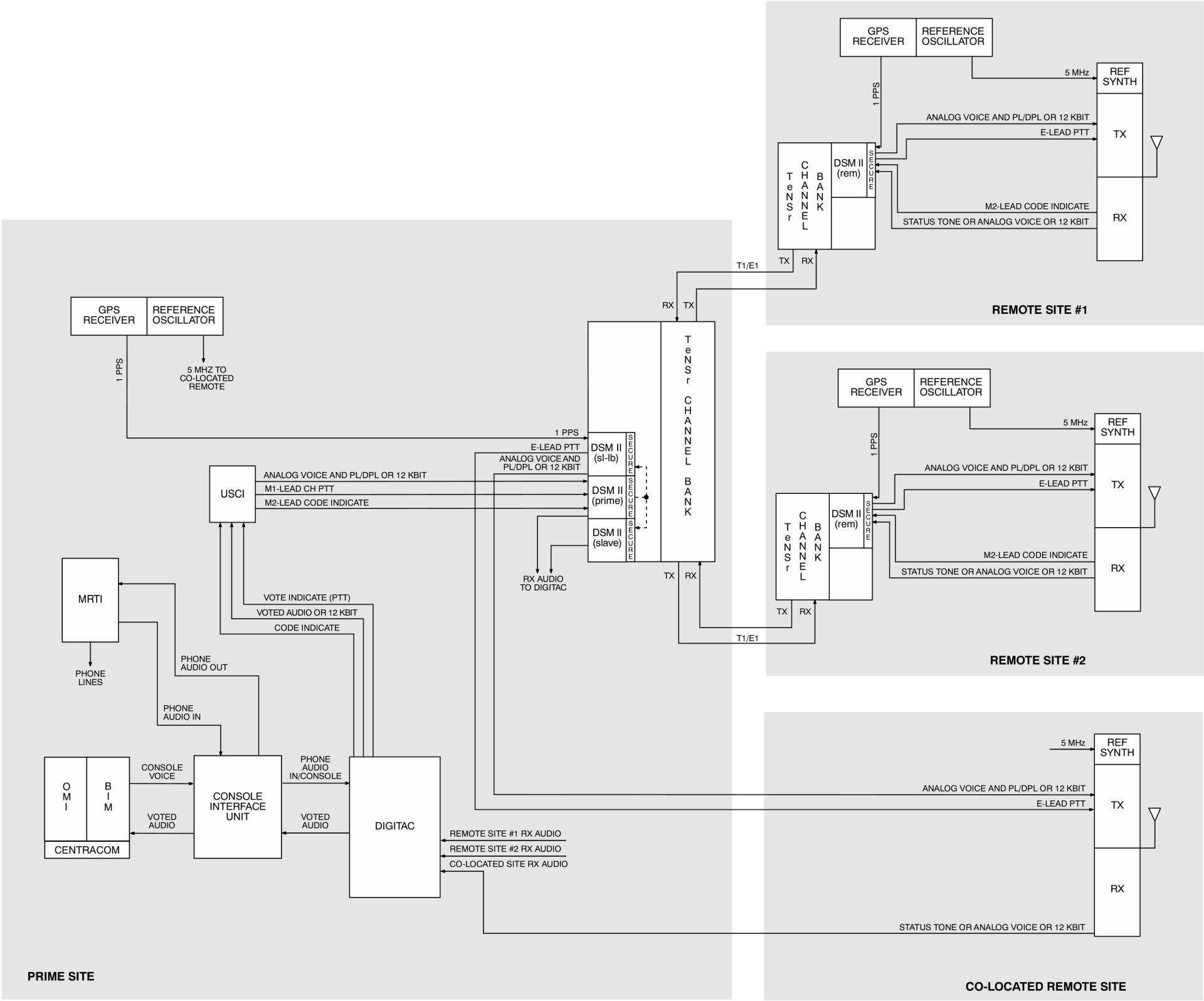
Figure 1-4 Conventional Digital Path Analog Clear Voice (Broadcast) With TeNSr Channel Banks





SIMU006A  
083000SKS

Figure 1-5 Trunked Digital Path Analog 12 kbit/Voice SecureNet (Broadcast) With TeNsR Channel Banks



SIMU048  
103196JNM

Figure 1-6 Conventional Digital Path Analog 12 kbit/Voice SecureNet (Broadcast) With TeNSr Channel Banks

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## Digital Mode

The digital-only simulcast configuration brings the key function of precise time synchronization of 9600 bps digital voice signals into the QUANTAR<sup>®</sup>/QUANTRO<sup>®</sup> station and the ASTRO-TAC/ASTRO-TAC 3000 comparator. In all other Motorola simulcast configurations such as dual-path analog, digital-path analog, and TeNSr analog, these functions are handled external to the station and comparator by specialized equipment. Figure 1-7 shows the simulcast time stamp operation used in the digital mode. Figures 1-8, 1-10, and 1-12 show trunked ASTRO only simulcast systems using TeNSr channel banks, Siemens channel banks, and analog microwave links, respectively. Figures 1-9, 1-11, and 1-13 show conventional ASTRO only simulcast systems using TeNSr channel banks, Siemens channel banks, and analog microwave links, respectively. Figures 1-14 and 1-15 show conventional ASTRO only simulcast systems using analog phone lines.

<b>NOTE</b>
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In Trunked systems, the control channel is configured identically to an analog voice channel. Therefore, in an ASTRO Trunked system where the control channel can be a voice channel, the channel will be configured as a mixed-mode channel.

### Conventional System Operation

In a conventional digital simulcast system, digital voice signals enter the comparator from the console via the digital interface unit (DIU) or from a receiver site. The ASTRO-TAC/ASTRO-TAC 3000 comparator has separate transmit and receive paths that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

The comparator embeds a digital time stamp in the outbound data stream every 180 msec (see Figure 1-7). This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the remote sites use to determine when to time launch the voice information onto the RF channel. The satellite time signal synchronizes the time-frequency reference of the prime site (comparator site) and all the remote transmitter sites to within 1  $\mu$ sec. The digital

## Digital Mode

audio signals are transported between the prime site (comparator site) and remote transmitter sites via V.24 links or internal wireline modems (conventional only). V.24 is the international equivalent to the U.S. EIA-232 (formerly RS-232) specification. These links are synchronous; the framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also use the request-to-send (RTS), clear-to-send (CTS) and carrier detect (CD) lines.

The V.24 links can be transported using many different types of equipment. TeNSr equipment with Sub-Rate Units (SRUs) can be used, V.3225 9600 bps modems can be used on an analog voice-grade circuit, or Siemens channel banks with Data Service Unit-Digital Path (DSU-DP) cards can be used to transport these V.24 signals.

ASTRO internal modems are commonly used, although are not required, for conventional voting and simulcast systems. These modems plug onto the comparator and station wireline boards as a daughterboard. These modems are not compatible with trunked systems; they require a voice-grade circuit. If the internal modems are not used in a digital system, a V.24 daughterboard is required.

## Trunked System Operation

In a trunked simulcast system, digital voice signals enter the comparator either from a receiver site or from a SmartZone audio switch.

The ASTRO-TAC/ASTRO-TAC 3000 comparator has separate transmit and receive paths that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC/ASTRO-TAC 3000 transmit process, digital voice signals enter the comparator either from the comparator receive process (system functioning in “site-trunking” mode) or from another site via the SmartZone audio switch (system functioning in “wide-trunking” mode). Next, the ASTRO-TAC/ASTRO-TAC 3000 comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the Data Broadcast Box (DBB). This embedded data (Link Control data) tells the radio that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

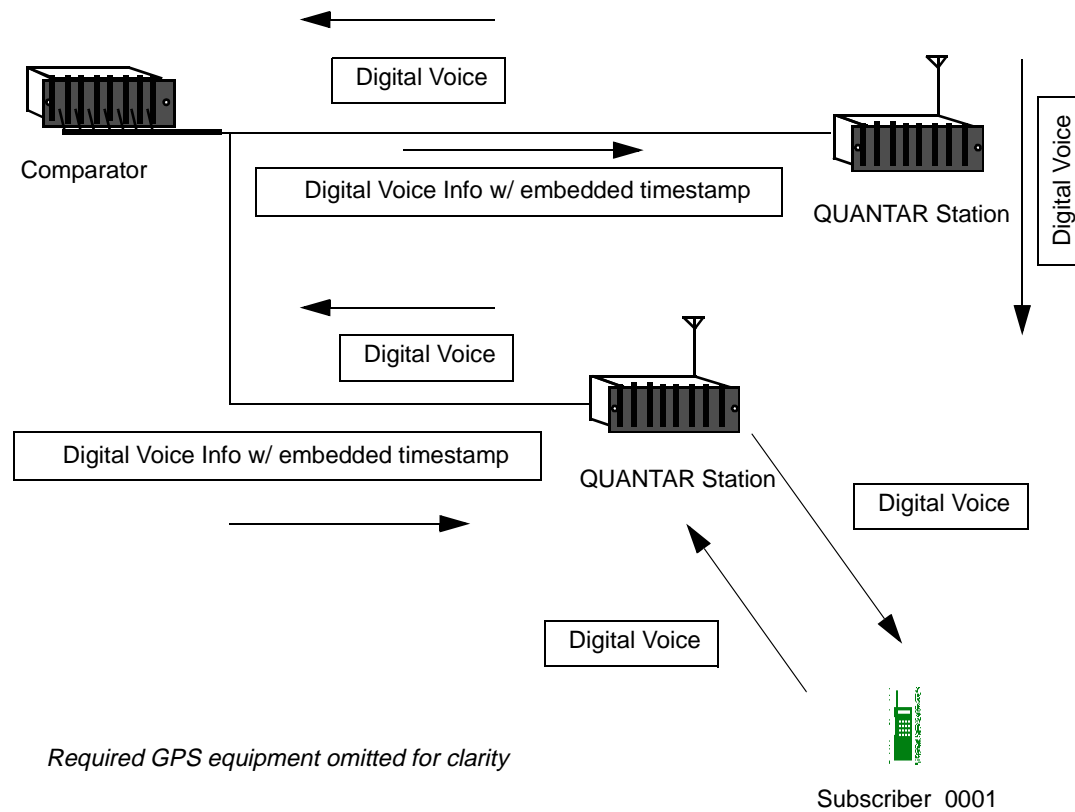
The ASTRO-TAC/ASTRO-TAC 3000 comparator embeds a digital time stamp in the outbound data stream every 180 msec (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

Trunked systems use analog channel equipment for transport of the control channel data. This means that either analog microwave (Starplus), Siemens (DSM), or TeNSr (DSM-II) equipment must be used for transport of the trunking control channel signals.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS and CD lines.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT (trunking PTT) line on the station parallel control cable. PTT also is signaled via the embedded data that the station receives with the digital voice information. Either of these methods will cause the station transmitter to key.

**Digital Mode**Figure 1-7 **Simulcast Timestamp Operation****Failsoft Operation**

Upon failure of key site components, the system reverts to failsoft operation so that users are still able to use the voice channels in a conventional repeat mode. In failsoft mode, all site transmitters are turned on and the embedded LC data is changed to a pattern that signals this condition to the subscribers. After sensing that the control channel is gone, the subscribers lock onto these failsoft voice channels.

- NOTES:
- 1. TRUNKED CONTROL CHANNEL IS ANALOG VIA DSM-II MODULES. TRUNKED CONTROL CHANNEL CONFIGURATION IS IDENTICAL TO THAT USED IN ANALOG VOICE CHANNEL (SEE FIGURE 1-1).
  - 2. THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

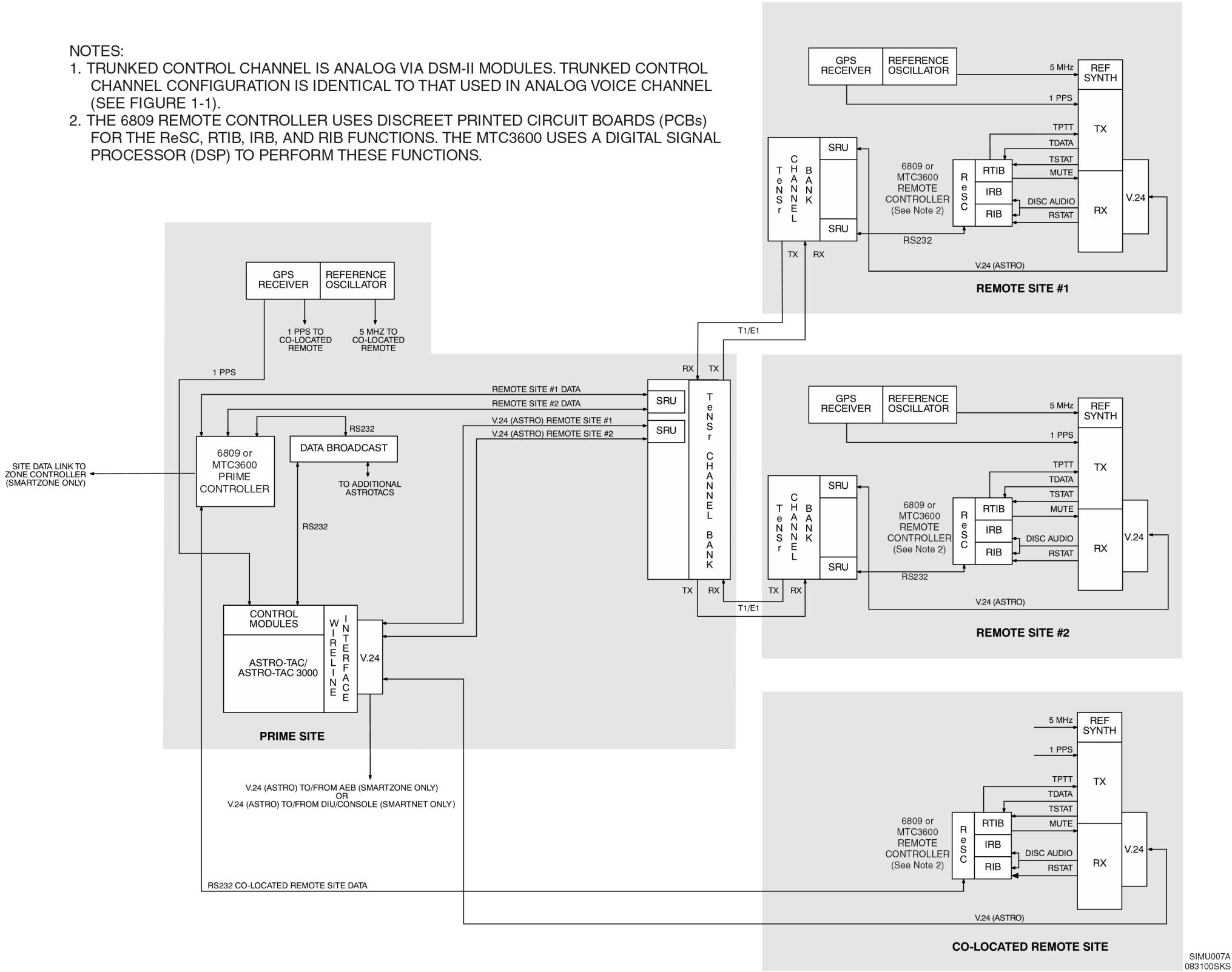


Figure 1-8 Trunked Digital Path ASTRO-Only Voice With TeNsR Channel Banks

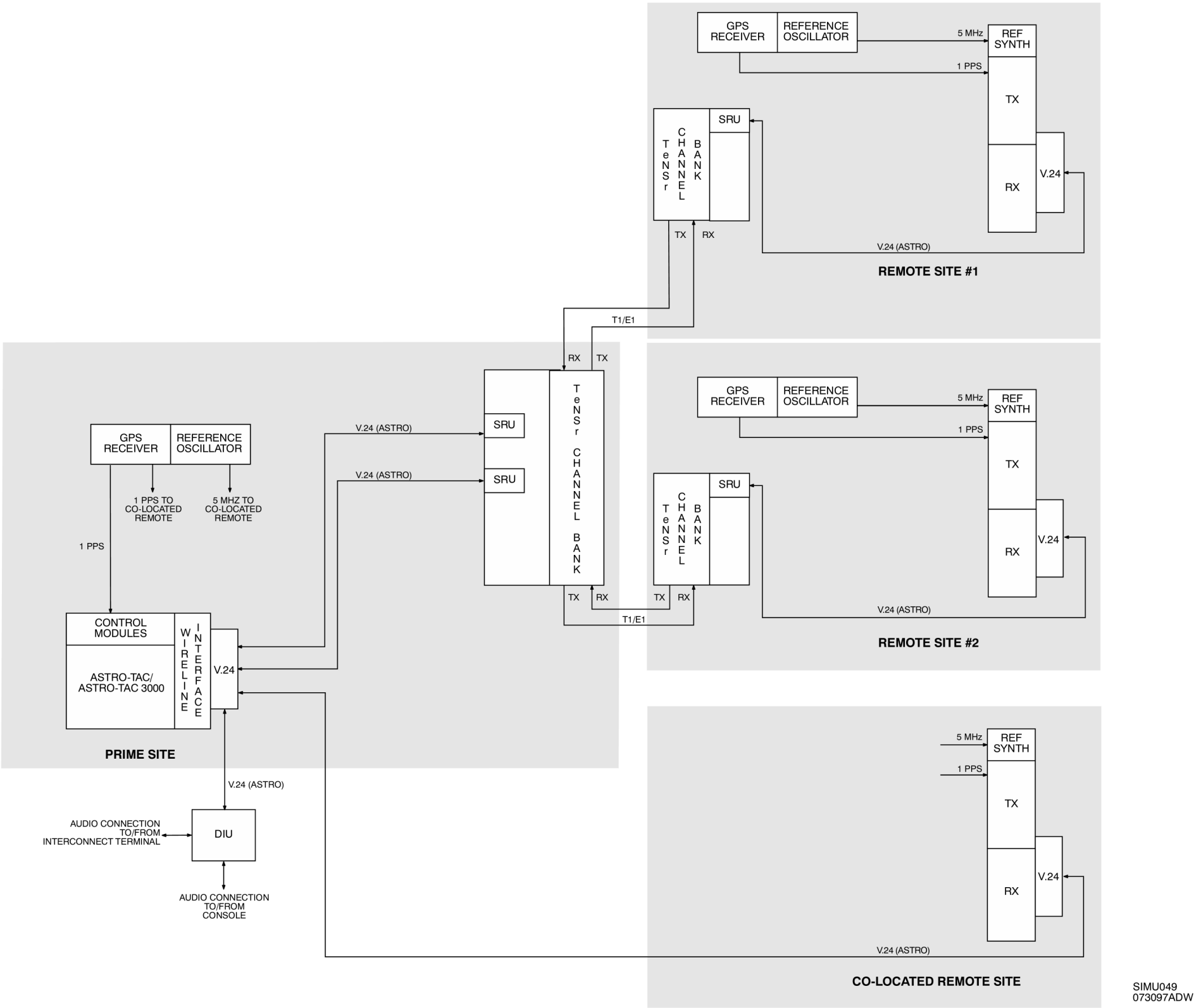


Figure 1-9 Conventional Digital Path ASTRO Only Voice With TeNSr Channel Banks



NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

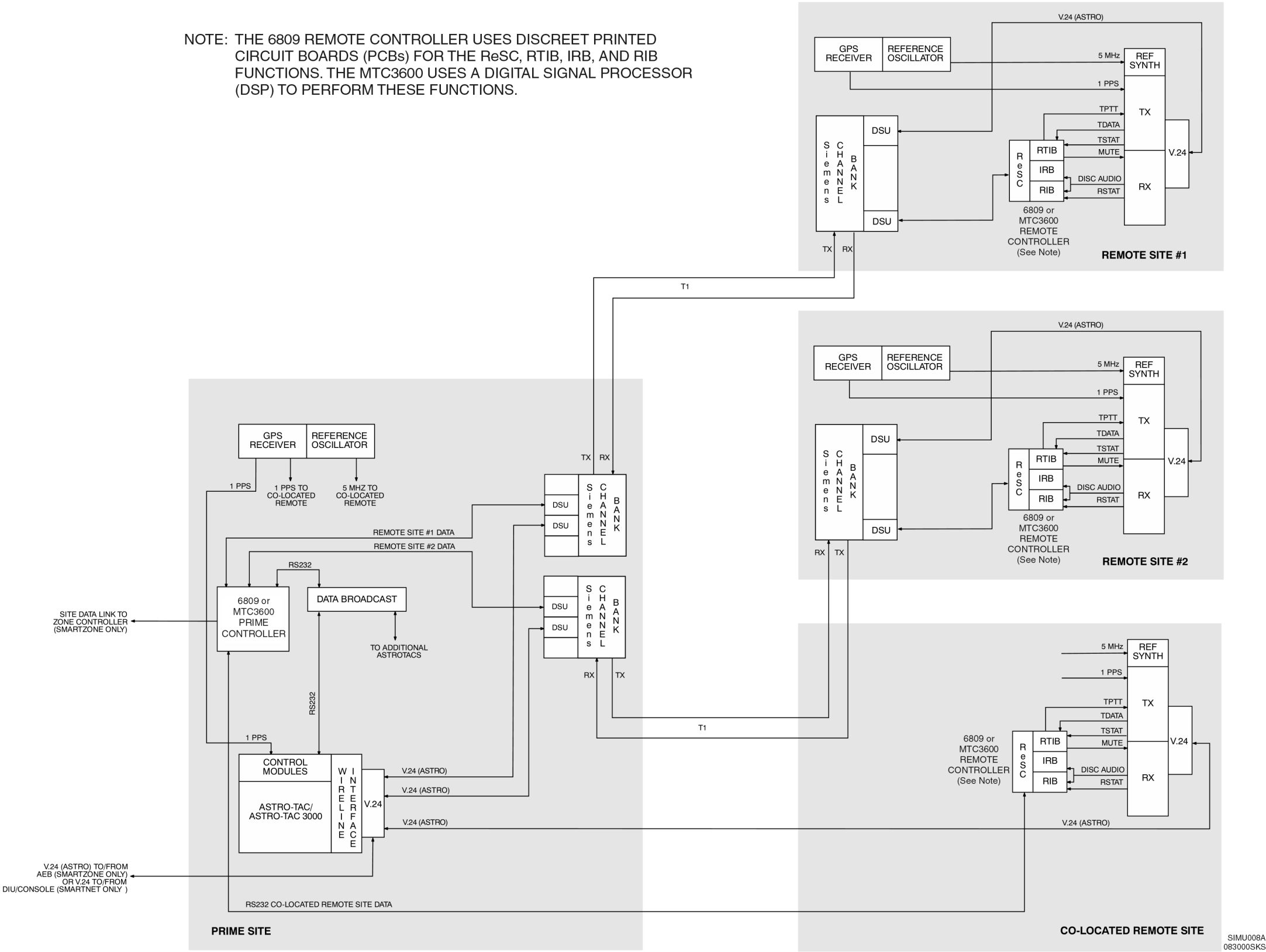
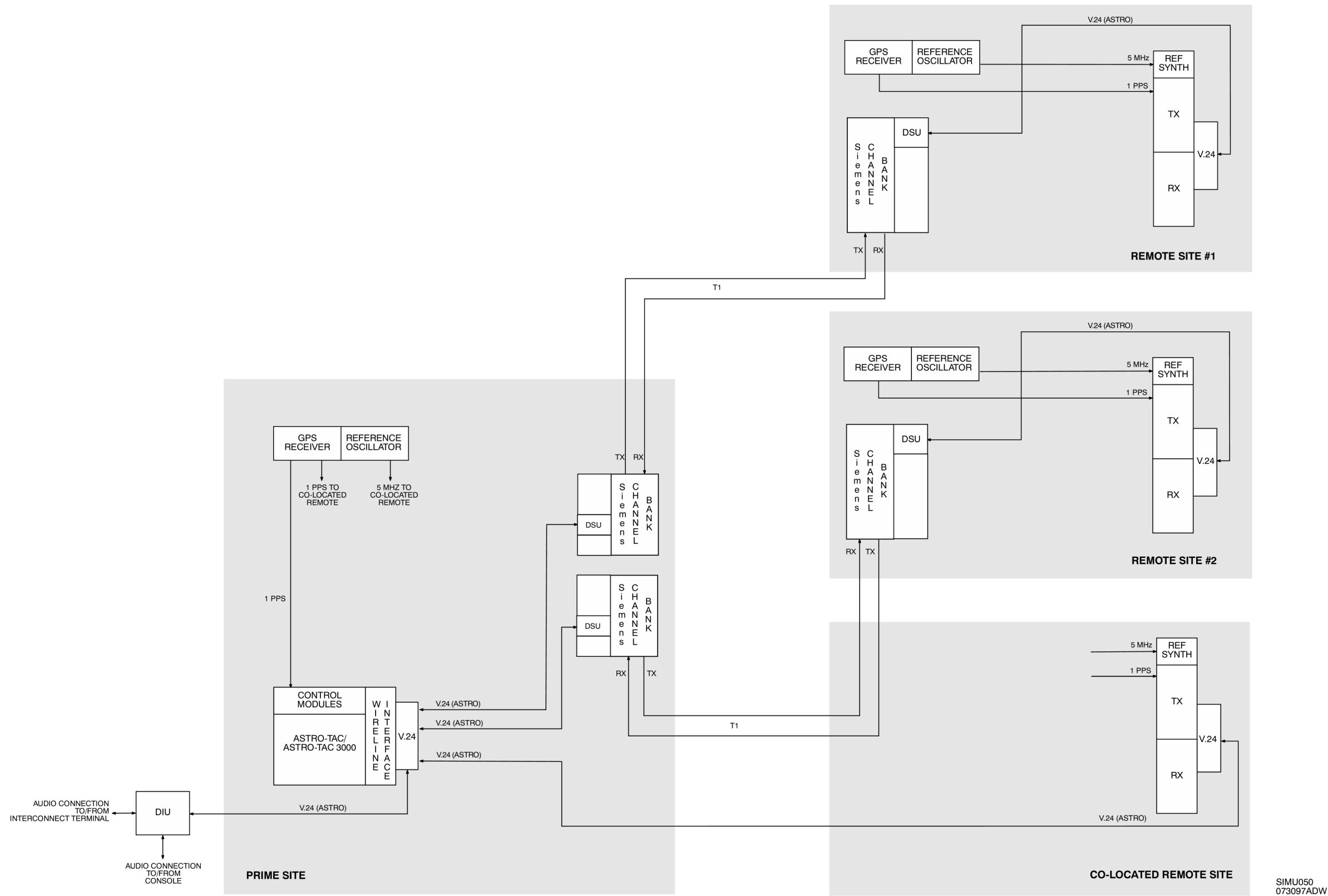


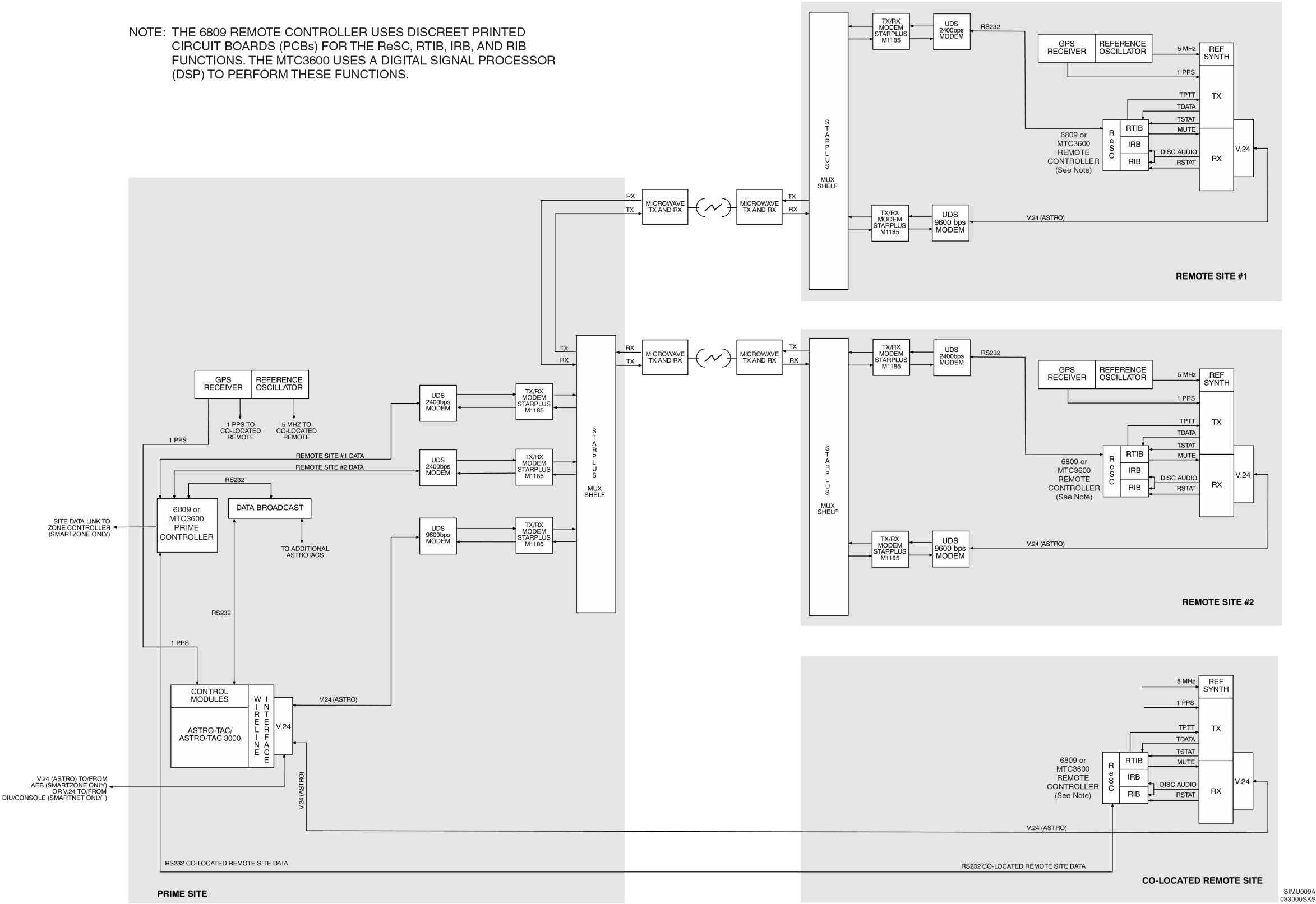
Figure 1-10 Trunked Digital Path ASTRO-Only Voice With Siemens Channel Banks

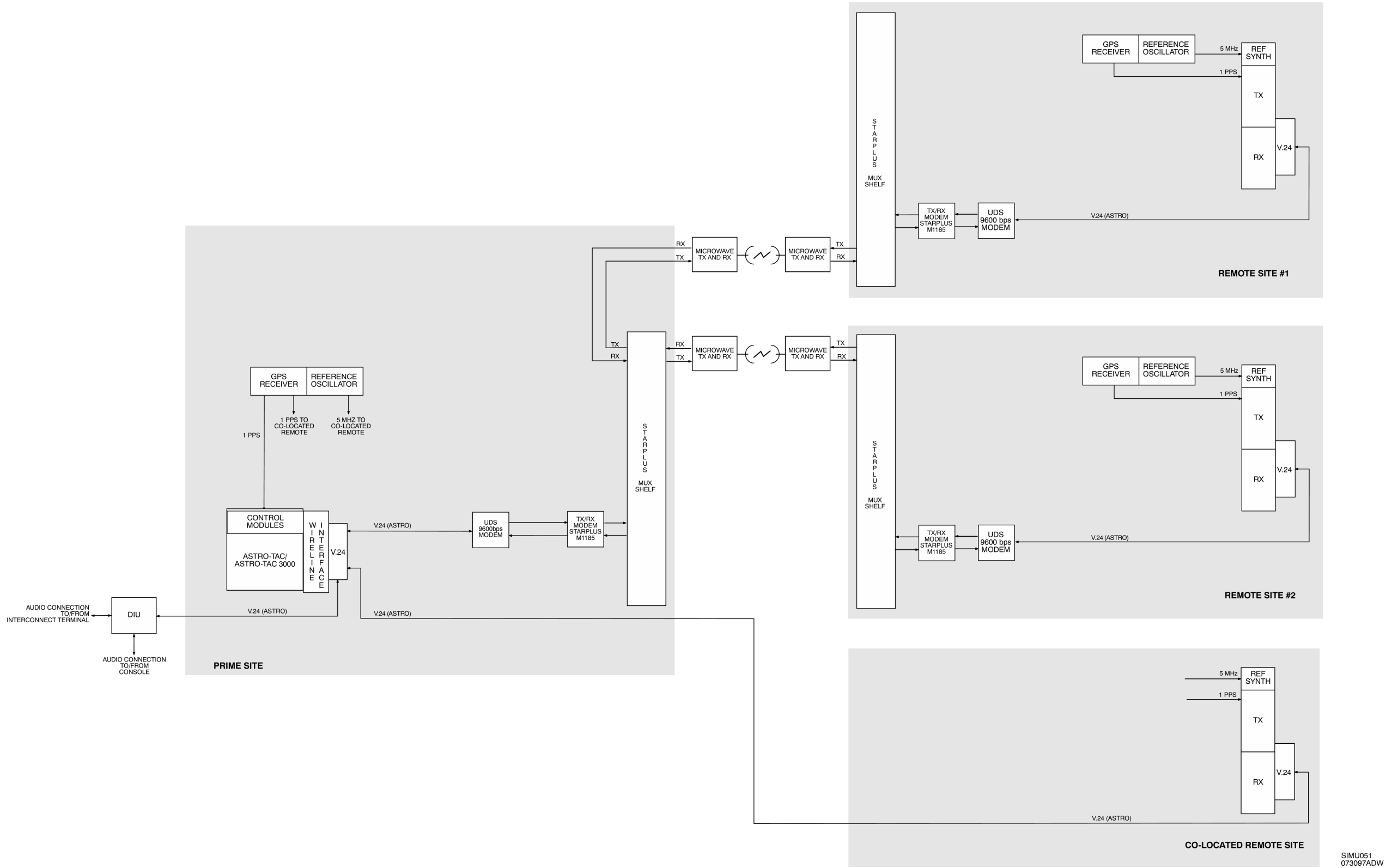


SIMU050  
073097ADW

Figure 1-11 Conventional Digital Path ASTRO-Only Voice With Siemens Channel Banks

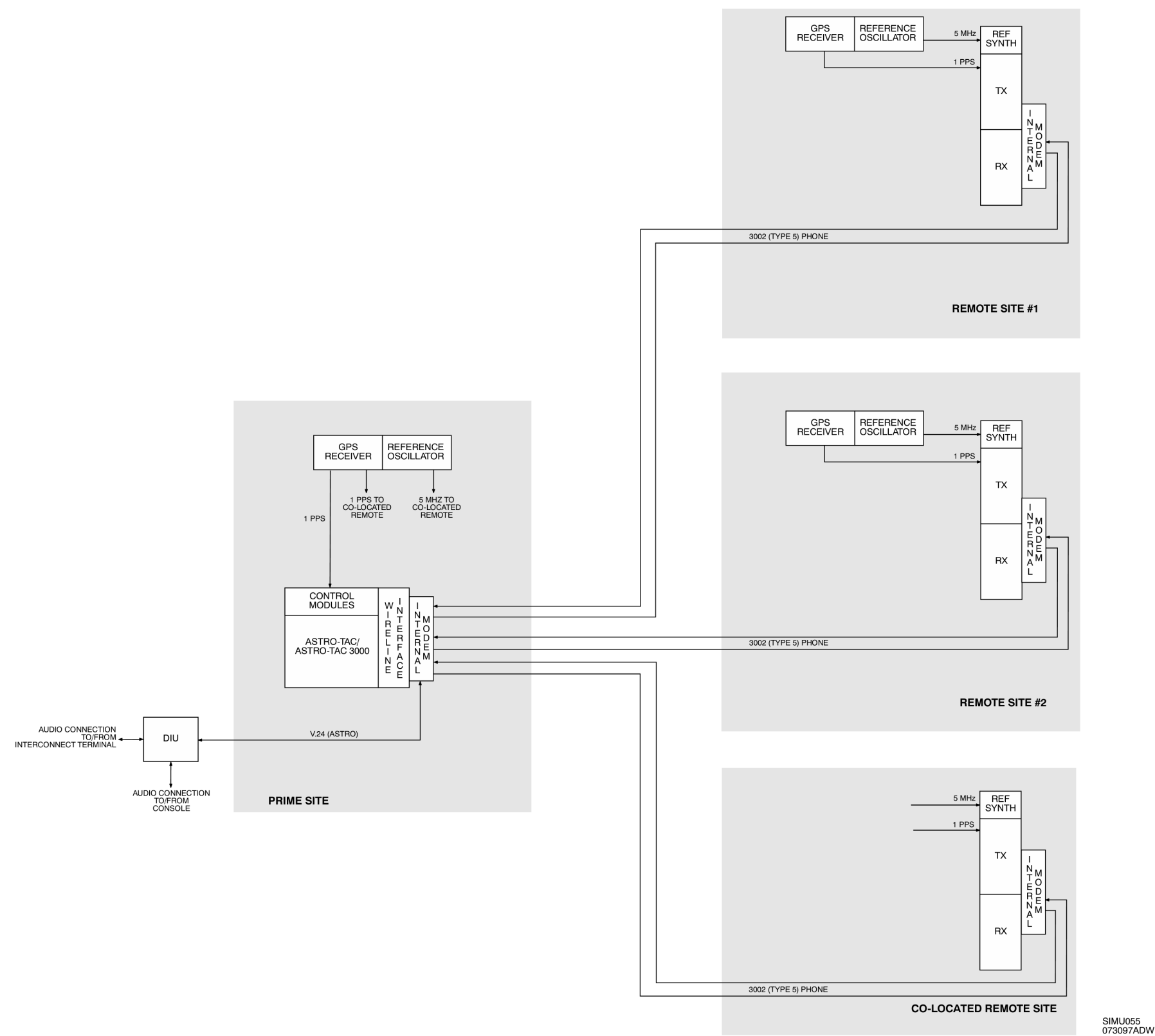
NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.





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Figure 1-13 Conventional Digital Path ASTRO-Only Voice With Analog Microwave Links



SIMU055  
073097ADW

Figure 1-14 Conventional Digital Path ASTRO-Only Voice With Analog Phone Lines (Using Internal Modems)

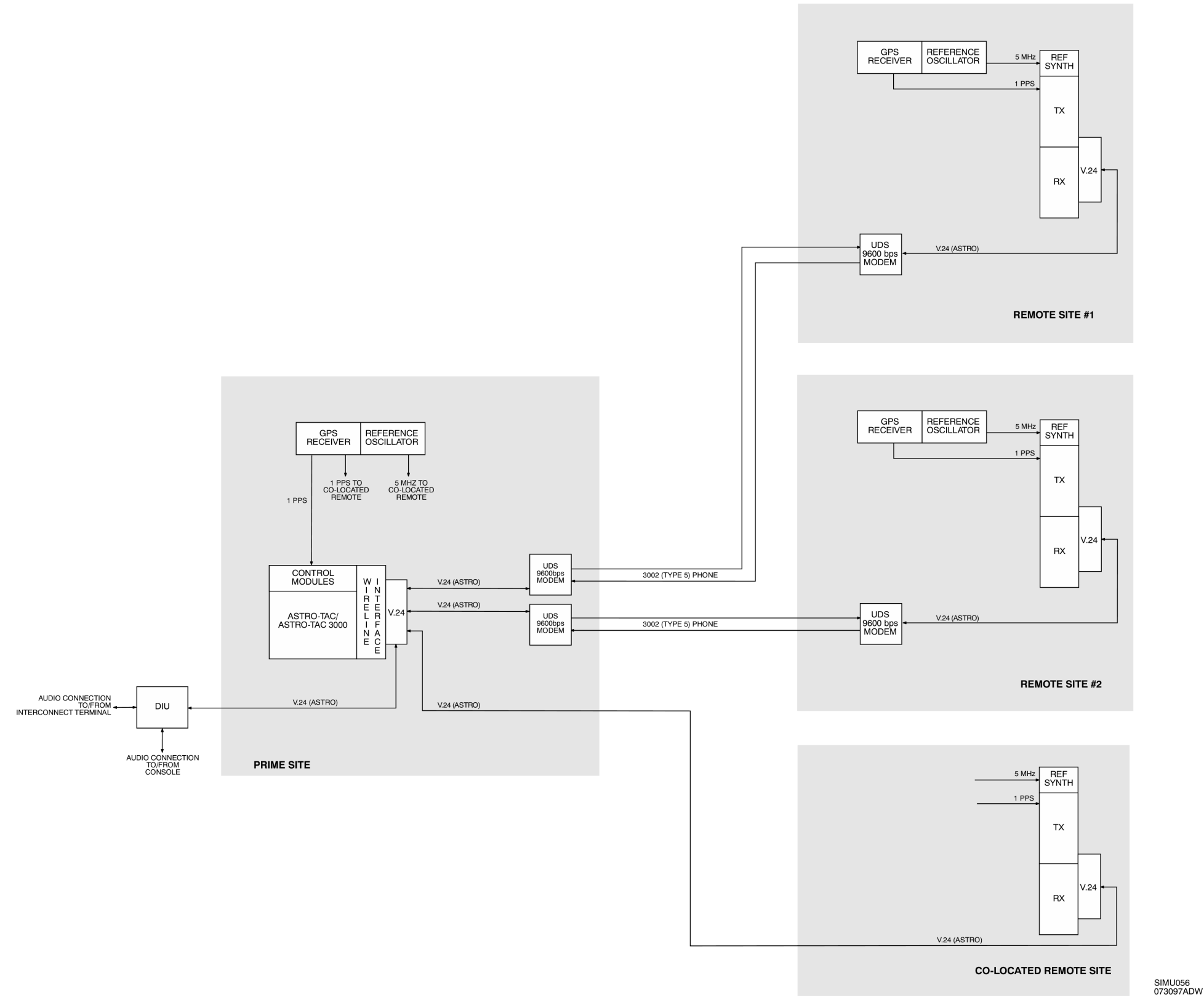


Figure 1-15 Conventional Digital Path ASTRO-Only Voice With Analog Phone Lines (Using External Modems)

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## Mixed Mode Using ASTRO-TAC

Figures 1-16 through 1-18 show mixed-mode trunked ASTRO-TAC simulcast systems using TeNSr channel banks, Siemens channel banks, and analog microwave links, respectively.

### Mixed Mode Using TeNSr Channel Banks

The TeNSr mixed-mode simulcast configuration uses signals from the GPS system to synchronize both analog and digital audio from multiple transmitters. By launching the audio at exactly the same time from multiple sites and carefully controlling the RF deviation and frequency, destructive interference of the transmitted signals is minimized. While the basic technique is the same for both the analog and digital signals, different equipment is used to synchronize and optimize the analog and digital audio signals (see Figure 1-16).

Between the comparator at the prime site and each of the remote site transmitters, there are separate digital and analog links. The digital and analog audio is transported on two separate time slots in the T1/E1 digital stream. The DSM-II card carries the analog and control channel information. The SRU card carries the digital voice information. SRU channels can be sub-rate multiplexed if desired. When channels are sub-rate multiplexed, up to five 9.6 kbps voice channel signals can be sent down a single 64 kbps time slot.

If the subscriber signal is analog, the prime site Digitac™ comparator votes the best quality analog audio and routes it to either the SmartZone Ambassador Electronics Bank (AEB) switch or through the USCI and back out to the simulcast transmitters. If the inbound subscriber signal is digital, the prime site ASTRO-TAC makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or to the simulcast transmitters. Signals that will travel to the AEB are combined into a single DS-0 by the TeNSr ADPCM, SRU and CPU cards prior to transmission over the wireline. This is done so that the AEB can switch both the digital and analog paths in a single DS-0.

The hybrid digital and analog links within a simulcast subsystem are always carried on separate time slots. The hybrid digital and analog links to a SmartZone master site are combined onto a single time slot by the TeNSr.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the DSM-II E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to key.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken

**Mixed Mode Using ASTRO-TAC**

data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.

**Digital Operation**

The ASTRO-TAC comparator has a separate transmit and receive path that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC transmit process, digital voice signals enter the comparator either from the comparator receive process (system in site-trunking) or from another site via the SmartZone audio switch (system in wide-trunking). Next, the ASTRO-TAC comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the DBB. This embedded data, called Link Control (LC) data, allows the radio to know that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC comparator embeds a digital time stamp in the outbound data stream every 180 ms (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to time launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS and CD lines. The SRU is used for transporting the V.24.

Trunked systems use “analog” channel equipment for transport of the control channel data. This means that TeNSr DSM-II equipment must be used for transport of the trunking control channel signals within a simulcast subsystem. With TeNSr channel banks, each channel bank can support up to eight T1s or E1s.



## Analog Operation

The Digitac comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

In the Digitac comparator transmit process, analog voice signals enter the Digitac comparator either directly from the receive process (system in wide or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the Digitac is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeed data. The USCI also generates failsoft data.

From the USCI, audio goes into the DSM-II card(s) in the TeNSr channel bank. The DSM-II then takes the audio and combines it with synchronization signals derived from the GPS 1 PPS reference. The signals reach the destination TeNSr equipment and the analog audio is transmitted out of the DSM-II (Digital Simulcast Modem, second generation) cards at precisely the same time and at a very similar amplitude. Since each station has slightly different gain characteristics, the DSM-II card output amplitude is adjusted during system optimization to precisely match station deviation over the air.

Refer to “Analog Mode Using DSM-IIIs,” page 1-8, for more information on analog operation using TeNSr channel banks.

### NOTE

Analog operation on a mixed-mode channel can use either the non-broadcast or broadcast mode.

## Failsoft Operation

During failsoft operation, the users are still able to use the voice channels in a conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals this condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeed data. After seeing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.

**Mixed Mode Using ASTRO-TAC****Mixed Mode Using Siemens Channel Banks**

In Figure 1-17, the Siemens digital path mixed-mode simulcast configuration uses signals from the GPS system to synchronize the digital audio from multiple transmitters. By launching the audio at exactly the same time from multiple sites and carefully controlling the RF deviation and frequency, destructive interference of the transmitted signals is minimized. The analog signals to each of the transmitters are not GPS synchronized, but are adjusted in phase and amplitude via adjustable delay/amplitude channel modems located at the prime site channel bank. A PC with special software talks to each of these devices via a data link to manually optimize each of the simulcast channels.

Between the comparator at the prime site and each of the remote site transmitters, separate time slots are used for the digital and analog links. Analog audio is carried over DSM cards. Digital audio is carried over V.24 links that are transported via DSU-DP data port cards.

If the subscriber signal is analog, the prime site Digitac comparator votes the best quality analog audio and routes it to either the SmartZone AEB switch or through the USCI and back out to the simulcast transmitters. If the inbound subscriber signal is digital, the prime site ASTRO-TAC makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or the simulcast transmitters. Signals that will travel to the AEB are eventually converted to a digital form and combined into a single DS-0 by the channel bank ADPCM and CPU cards. In most SmartZone systems, this is done at the SmartZone master site. This is required so that the AEB can switch both the digital and analog paths in a single DS-0.

The hybrid digital and analog links within a simulcast subsystem are always carried on separate digital carrier timeslots. The hybrid digital and analog links to a SmartZone master must be combined onto single timeslot prior to entering the SmartZone audio switch.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the DSM E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to activate.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.

## Digital Operation

The ASTRO-TAC comparator has a separate transmit and receive path that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC transmit process, digital voice signals enter the comparator either from the comparator receive process (system in wide- or site-trunking) or from another site via the SmartZone audio switch (system in wide-trunking). Next, the ASTRO-TAC comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the data broadcast box. This embedded data, called Link Control (LC) data, allows the radio to know that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC comparator embeds a digital time stamp in the outbound data stream every 180 ms (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to time launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS, and CD lines. In the digital path Siemens transmission system, the 9600 bps synchronous V.24 links are connected between prime and remote sites by using DSU-DP dataport cards.

Trunked systems use analog channel equipment for transport of the control channel data. This means that DSM cards must be used for transport of the trunking control channel signals within a simulcast subsystem.

## Analog Operation

The Digitac comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

**Mixed Mode Using ASTRO-TAC**

In the Digitac comparator transmit process, analog voice signals enter the Digitac comparator either directly from the receive process (system in wide- or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the Digitac is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeed data. The USCI also generates failsoft data.

The USCI audio and high-speed output routes through the DSMs to the remote sites. The microwave channels are not party-lined; therefore, each channel for each site requires one transmit and receive modem pair. Since the USCI is designed to drive only one 600 $\Omega$  load, the SDA multiplexes the required number of transmit modems for the required number of simulcast sites. For every remote site, there is a corresponding channel bank and Simulcast Serial Adapter (SSA) at the prime site. Each of the SSAs communicates with the DSMs located in the channel banks at the prime site.

At the remote site, the DSM RX modem output routes the audio and data to the base station. (Digital Path systems do not require an Remote Delay Module (RDM) since this is clear-only analog.) The delay and amplitude can be set remotely using the SSA and the Prime Optimization Node (PON).

The PTT signal in a Digital Path system originates in the prime site central controller during trunking operation and is sent to the USCI. During failsoft, PTT originates at the USCI and is routed to the M-lead input of the DSM. The E-lead output of the receiving modem is connected to the DSM PTT input and routed to the repeater PTT input. PTT begins at the prime site central controller, is routed from the USCI through the SDA distribution amplifier to the channel bank and DSMs, and on to the repeater.

The receive path of the Digital Path simulcast system begins with the base station repeater. The inbound RF signal from the subscriber radio is received at the repeater and demodulated. The signal is then routed to the transmit line of the DSM for transmission to the prime site. The DSM converts the received audio voltage to digital samples, which are applied to the transmit line interface units Pulse Code Modulation (PCM) bus in the channel bank. The transmit line interface unit places the digital samples from the DSM in the channel bank into its assigned time slot in the resulting DS-1 Time Division Multiplex (TDM) digital signal. The DS-1 signal is applied to the input of the digital microwave radio or telco facility. At the prime site, the received DS-1 line interface unit demultiplexes the DS-1 and delivers the appropriate digital signal to the DSM corresponding to the sending DSM at the remote site. The demodulator function of the DSM restores the digit samples to voltages which are then filtered to produce the desired audio signal frequencies. The receive audio signal is routed to the receiving Digitac voting comparator. The comparator selects the best signal based on the highest signal-to-noise ratio. This signal, called the voted audio output, is then sent to the USCI if the system is in failsoft mode, site trunking or stand-alone simulcast. If the simulcast system is part of a SmartZone system, the voted audio is sent up to the master site.

## Failsoft Operation

During failsoft operation, the users are still able to use the voice channels in a conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals this condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeed data. After seeing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.

## Mixed Mode Using Analog Microwave

In Figure 1-18, the analog microwave mixed-mode simulcast configuration uses signals from the GPS system to synchronize the digital audio from multiple transmitters.

Between the comparator at the prime site and each of the remote site transmitters, there are separate digital and analog links. The digital and analog audio is transported on two separate voice channels in the microwave multiplex system. Analog audio is carried over special wideband analog voice cards. Analog audio is party-lined for the distribution. Digital audio is carried over Single-Sideband (SSB) voice cards using 9600 bps synchronous modems.

If the subscriber signal is analog, the prime site Digitac comparator votes the best quality analog audio and routes it to either the SmartZone AEB switch or back to the simulcast transmitters. If the inbound subscriber signal is digital, the prime site ASTRO-TAC makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or through the USCI, and then back to the simulcast transmitters. Signals that will travel to the AEB are eventually converted to a digital form and combined into a single DS-0 signal by the channel bank ADPCM, SRU and CPU cards. In most SmartZone analog microwave systems, this is done at the SmartZone prime site. This is required so that the AEB can switch both the digital and analog paths in a single DS-0 signal.

Note that the hybrid digital and analog links within a simulcast subsystem are always carried on separate microwave mux voice channels. The hybrid digital and analog links to a SmartZone Prime site must be combined onto a single time slot prior to entering the SmartZone audio switch.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the wide band modem's E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to key.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is

**Mixed Mode Using ASTRO-TAC**

present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.

**Digital Operation**

The ASTRO-TAC comparator has separate transmit and receive paths that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC transmit process, digital voice signals enter the comparator either from the comparator receive process (system functioning in site-trunking mode) or from another site via the SmartZone audio switch (system functioning in wide-trunking mode). Next, the ASTRO-TAC comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the DBB. This embedded data Link Control (LC) data tells the radio that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC comparator embeds a digital time stamp in the outbound data stream every 180 msec (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS, and CD lines.

In the dual path analog microwave system, the V.24 links are typically connected between prime and remote sites using V.3225, 9600 bps modems. Each of these modems is then connected to a normal 4 kHz SSB duplex voice card in the analog microwave mux shelf.

Trunked systems use analog channel equipment for transport of the control channel data. This means that wide-band modems must be used for transport of the trunking control channel signals within a simulcast subsystem.

## Analog Operation

The Digitac comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

In the Digitac comparator transmit process, analog voice signals enter the Digitac comparator either directly from the receive process (system in wide- or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the Digitac is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeed data. The USCI also generates failsoft data.

From the USCI, audio and high-speed data are sent down one path, while lowspeed, failsoft and disconnect data are Frequency Shift Keying (FSK) encoded and sent down the second path. The USCI also generates failsoft data.

The USCI audio and high-speed output go to a single Starplus transmit-only wideband modem, while the FSK output routes to a single Starplus transmit-only single sideband modem. The audio and high-speed path us wideband modems because high-speed data requires a bandwidth greater than single sideband modems can provide. Also, the wideband modems have linear phase response which allows for audio phasing by adding linear delay to the audio path.

Both the wideband and SSB multiplex paths are set up in a party-line configuration. This configuration uses a single transmitting modem with multiple receiving modems. This design reduces the total baseband loading by reducing the number of carriers on the microwave baseband.

At the remote site, the receiving multiplex modems provide the audio, high-speed, and FSK inputs to the Remote Delay Module (RDM). The RDM converts the FSK to lowspeed data and provides amplitude and linear delay adjustment. The RDM also provides these same adjustments separately for the audio and high-speed via the (Primary Optimization Node (PON). From the PON at the prime site, the installer addresses a specific RDM, enters the amplitude and delay values, and then transmits them over a data link to the remote site. The delay adjustment on the RDM allows all transmitters on a single frequency to transmit all signals in phase within  $\pm 2.6 \mu\text{sec}$ .

Transmitting all signals in phase prevents one form of simulcast distortion. The amplitude adjustment on the RDM allows the audio, high-speed, and low-speed data (which includes failsoft and disconnect data) to be adjusted separately in 0.05 dB increments. This causes all transmitters on a single frequency to have the same deviation, which prevents another form of simulcast distortion. This fine-tuning adjustment compensates for level errors that may have occurred at one or more of the many adjustment points in the repeat audio path.

**Mixed Mode Using ASTRO-TAC**

After the delay and amplitude adjustments occur on the RDM, the audio, highspeed and lowspeed data paths are summed in the RDM. The single output of the RDM is connected to the base station. At the base station, the audio/data is fed to the base station's wideband input.

For the analog portion of a mixed-mode channel using analog microwave, the PTT signal originates in the prime site central controller during trunking operation and is sent to the USCI. During failsoft, PTT originates at the USCI and is routed to the M-lead of the transmit only wideband multiplex modem. The E-lead output of the receiving modem is connected to the RDM PTT input and the RDM PTT output is connected to the base station PTT input. PTT is routed from the USCI through the multiplexer, through the RDM, and on to the base station.

The receive path begins at the point where the simulcast base station receives RF signals and ends at the prime site's voting comparator. The inbound radio subscriber RF signals are received at the simulcast base station and demodulated. The audio portion of the signal is routed to a Starplus transmit-only SSB modem. The Starplus modem sends the audio signals to a receive-only SSB modem at the prime site. From the multiplex modem, the audio signals are routed to a Digitac voting comparator. The comparator selects the best signal based on the highest signal-to-noise ratio. This signal, called the voted audio output, is then sent to the USCI if the system is in failsoft mode, site trunking or stand-alone simulcast. If the simulcast system is part of a SmartZone system, the voted audio is sent up to the prime site. The trunking control channel is treated and handled as an analog voice channel.

**Failsoft Operation**

During failsoft operation, the users are still able to use the voice channels in conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals the failsoft condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeed data. After sensing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.



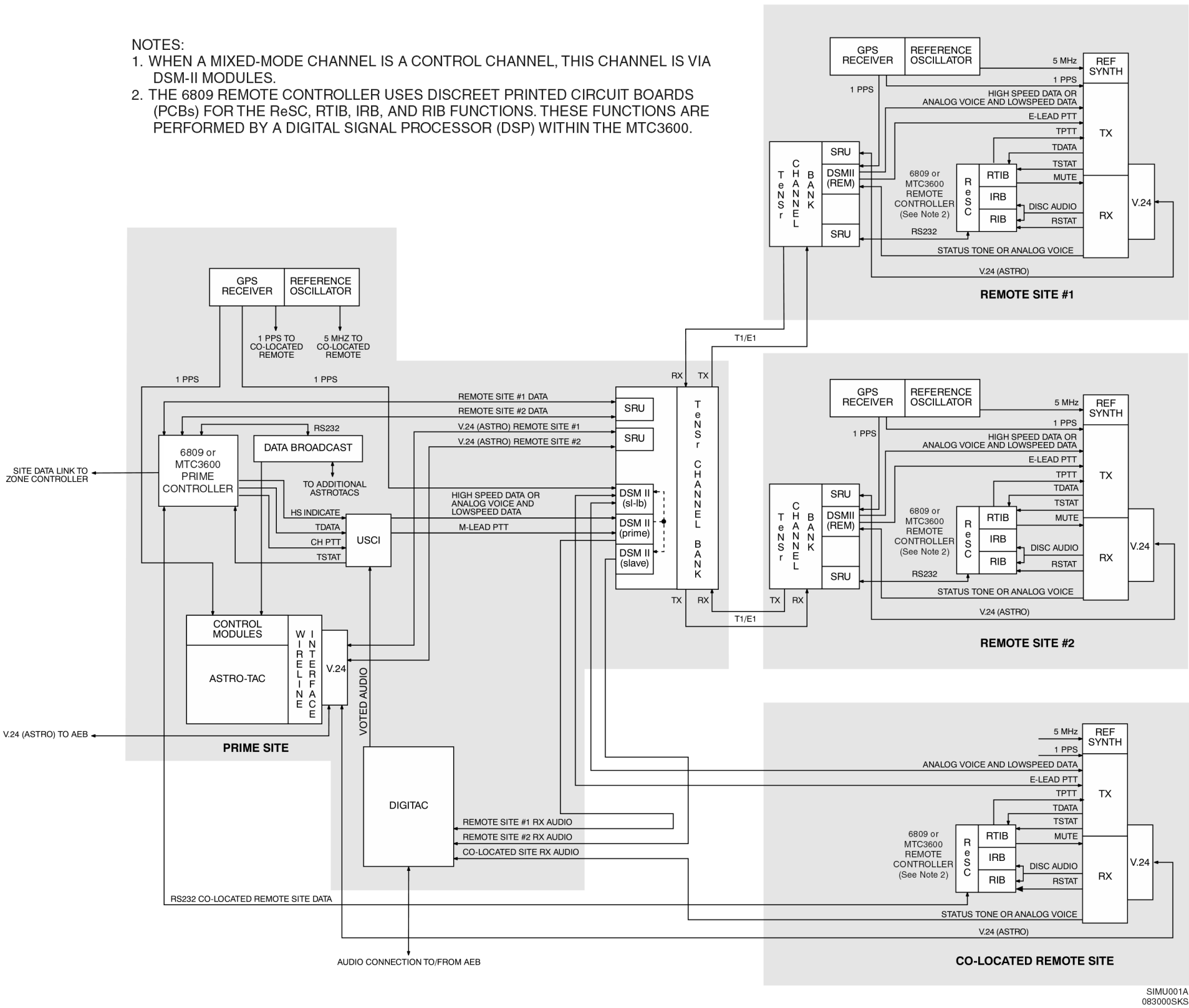


Figure 1-16 Digital Path ASTRO Mixed Mode With TeNSr Channel Banks (SmartZone Subsystem)

NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

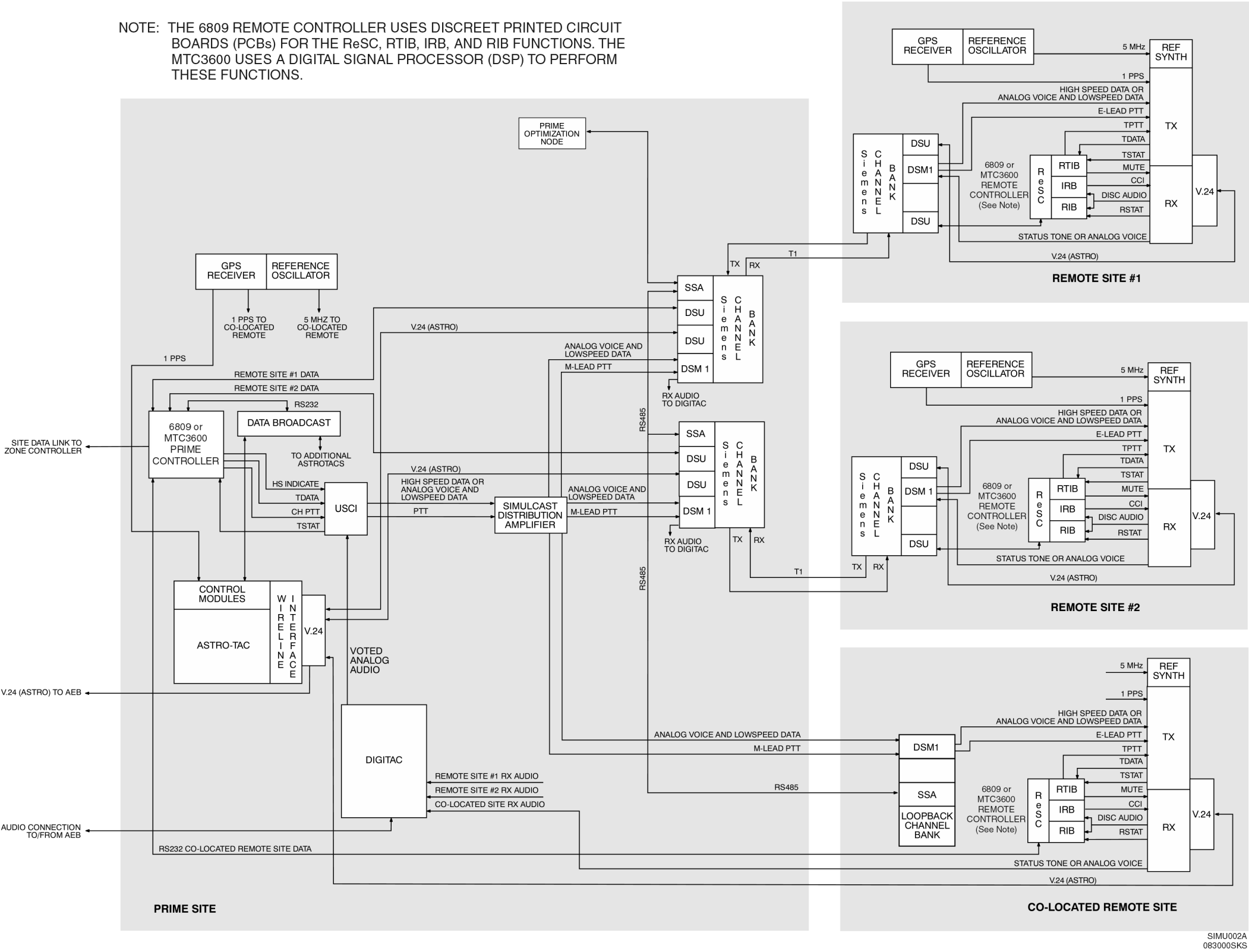


Figure 1-17 Digital Path ASTRO Mixed Mode With Siemens Channel Banks (SmartZone Subsystem)

NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

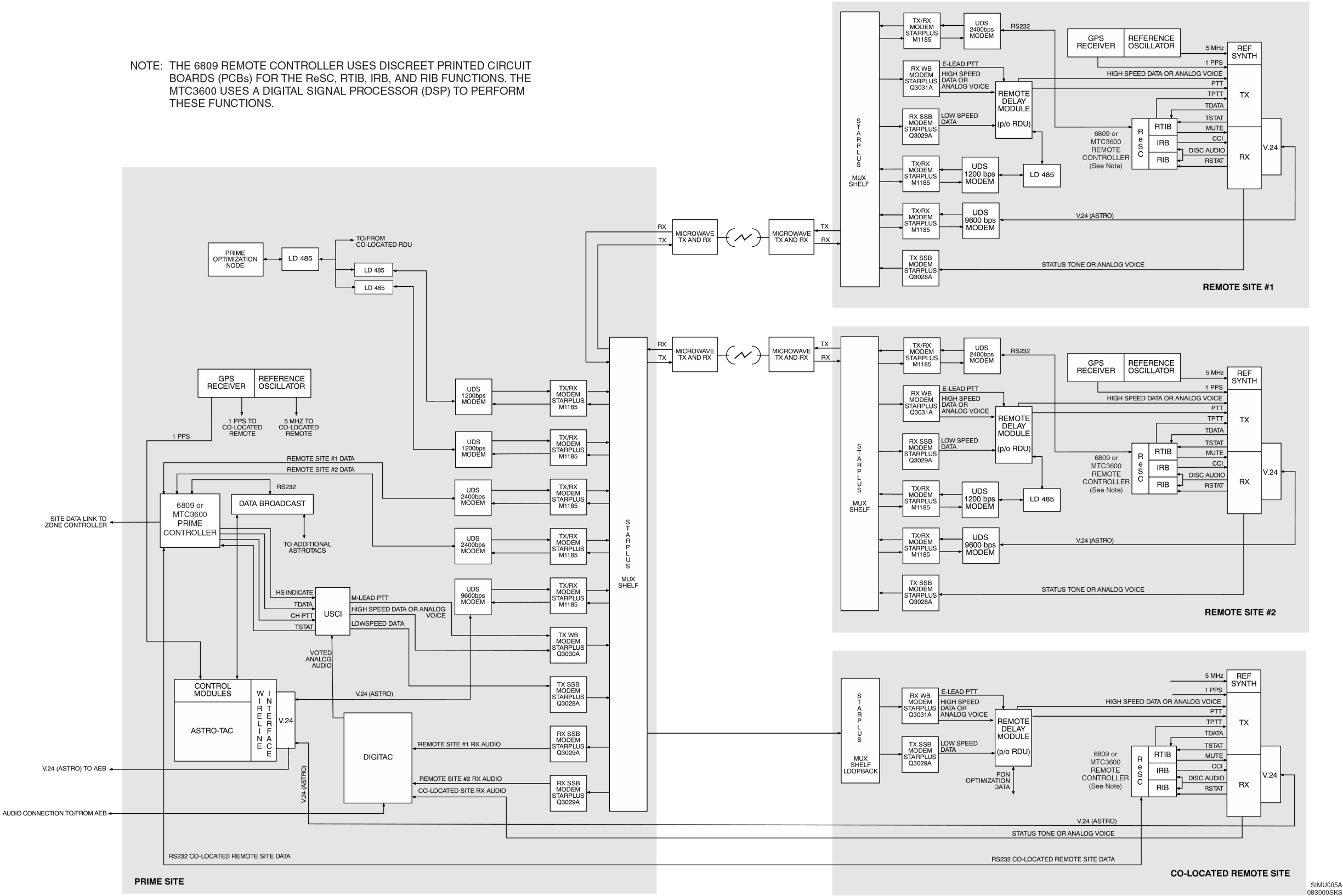


Figure 1-18 ASTRO Mixed Mode With Analog Microwave (SmartZone Subsystem)

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## Mixed Mode Using ASTRO-TAC 3000

The ASTRO-TAC 3000 comparator combines the functions of the ASTRO-TAC and Digitac comparators (discussed in “Mixed Mode Using ASTRO-TAC,” in one unit. Figures 1-19 through 1-21 show mixed-mode trunked ASTRO-TAC 3000 simulcast systems using TeNSr channel banks, Siemens channel banks, and analog microwave links, respectively.

### Mixed Mode Using TeNSr Channel Banks

The TeNSr mixed-mode simulcast configuration uses signals from the GPS system to synchronize both analog and digital audio from multiple transmitters (see Figure 1-19). By launching the audio at exactly the same time from multiple sites and carefully controlling the RF deviation and frequency, destructive interference of the transmitted signals is minimized. While the basic technique is the same for both the analog and digital signals, different equipment is used to synchronize and optimize the analog and digital audio signals.

Between the comparator at the prime site and each of the remote site transmitters, there are separate digital and analog links. The digital and analog audio is transported on two separate time slots in the T1/E1 digital stream. The DSM-II card carries the analog and control channel information. The SRU card carries the digital voice information. SRU channels can be sub-rate multiplexed if desired. When channels are sub-rate multiplexed, up to five 9.6 kbps voice channel signals can be sent down a single 64 kbps time slot.

If the subscriber signal is analog, the ASTRO-TAC 3000 comparator at the prime site votes the best quality analog audio and routes it to either the SmartZone AEB switch or through the USCI and back out to the simulcast transmitters. If the inbound subscriber signal is digital, the ASTRO-TAC 3000 makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or to the simulcast transmitters. Signals that will travel to the AEB are combined into a single DS-0 by the TeNSr ADPCM, SRU and CPU cards prior to transmission over the wireline. This is done so that the AEB can switch both the digital and analog paths in a single DS-0.

The hybrid digital and analog links within a simulcast sub-system are always carried on separate time slots. The hybrid digital and analog links to a SmartZone master site are combined onto a single time slot by the TeNSr.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the DSM-II E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to key.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is

**Mixed Mode Using ASTRO-TAC 3000**

present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.

**Digital Operation**

The ASTRO-TAC 3000 comparator has a separate transmit and receive path that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC 3000 transmit process, digital voice signals enter the comparator either from the comparator receive process (system in site-trunking) or from another site via the SmartZone audio switch (system in wide-trunking). Next, the ASTRO-TAC 3000 comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the DBB. This embedded data, called Link Control (LC) data, allows the radio to know that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC 3000 comparator embeds a digital time stamp in the outbound data stream every 180 ms (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to time launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS and CD lines. The SRU is used for transporting the V.24.

Trunked systems use “analog” channel equipment for transport of the control channel data. This means that TeNSr DSM-II equipment must be used for transport of the trunking control channel signals within a simulcast sub-system. With TeNSr channel banks, each channel bank can support up to eight T1s or E1s.

## Analog Operation

The ASTRO-TAC 3000 comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

In the ASTRO-TAC 3000 comparator transmit process, analog voice signals enter the ASTRO-TAC 3000 comparator either directly from the receive process (system in wide- or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the ASTRO-TAC 3000 is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeak data. The USCI also generates failsoft data.

From the USCI, audio goes into the DSM-II (digital simulcast modem) card(s) in the TeNSr channel bank. The DSM-II then takes the audio and combines it with synchronization signals derived from the GPS 1 PPS reference. The signals reach the destination TeNSr equipment and the analog audio is transmitted out of the DSM-II cards at precisely the same time and at a very similar amplitude. Since each station has slightly different gain characteristics, the DSM-II card output amplitude is adjusted during system optimization to precisely match station deviation over the air.

Refer to the “Analog Mode Using DSM-II’s” section of this chapter for more information on analog operation using TeNSr channel banks.

### NOTE

Analog operation on a mixed-mode channel can use either the non-broadcast or broadcast mode.

## Failsoft Operation

During failsoft operation, the users are still able to use the voice channels in a conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals this condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeak data. After seeing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC 3000 comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.

**Mixed Mode Using ASTRO-TAC 3000****Mixed Mode Using Siemens Channel Banks**

The Siemens digital path mixed-mode simulcast configuration uses signals from the GPS system to synchronize the digital audio from multiple transmitters (see Figure 1-20). By “launching” the audio at exactly the same time from multiple sites and carefully controlling the RF deviation and frequency, destructive interference of the transmitted signals is minimized. The analog signals to each of the transmitters are not GPS synchronized, but are adjusted in phase and amplitude via adjustable delay/amplitude channel modems located at the prime site channel bank. A PC with special software talks to each of these devices via a data link to manually optimize each of the simulcast channels.

Between the comparator at the prime site and each of the remote site transmitters, separate time slots are used for the digital and analog links. Analog audio is carried over DSM cards. Digital audio is carried over V.24 links that are transported via DSU-DP data port cards.

If the subscriber signal is analog, the ASTRO-TAC 3000 comparator votes the best quality analog audio and routes it to either the SmartZone AEB switch or through the USCI (Universal Simulcast Controller Interface) and back out to the simulcast transmitters. If the inbound subscriber signal is digital, the prime site ASTRO-TAC 3000 makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or the simulcast transmitters. Signals that will travel to the AEB are eventually converted to a digital form and combined into a single DS-0 by the channel bank ADPCM and CPU cards. In most SmartZone systems, this is done at the SmartZone master site. This is required so that the AEB can switch both the digital and analog paths in a single DS-0.

The hybrid digital and analog links within a simulcast sub-system are always carried on separate digital carrier timeslots. The hybrid digital and analog links to a SmartZone master must be combined onto single timeslot on prior to entering the SmartZone audio switch.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the DSM E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to activate.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.



## Digital Operation

The ASTRO-TAC 3000 comparator has a separate transmit and receive path that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC 3000 transmit process, digital voice signals enter the comparator either from the comparator receive process (system in wide or site-trunking) or from another site via the SmartZone audio switch (system in wide-trunking). Next, the ASTRO-TAC 3000 comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the data broadcast box. This embedded data, called Link Control (LC) data, allows the radio to know that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC 3000 comparator embeds a digital time stamp in the outbound data stream every 180 ms (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to time launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS, and CD lines. In the digital path Siemens transmission system, the 9600 bps synchronous V.24 links are connected between prime and remote sites by using DSU-DP dataport cards.

Trunked systems use “analog” channel equipment for transport of the control channel data. This means that DSM cards must be used for transport of the trunking control channel signals within a simulcast sub-system.

## Analog Operation

The ASTRO-TAC 3000 comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

**Mixed Mode Using ASTRO-TAC 3000**

In the ASTRO-TAC 3000 comparator transmit process, analog voice signals enter the ASTRO-TAC 3000 comparator either directly from the receive process (system in wide or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the ASTRO-TAC 3000 is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeed data. The USCI also generates failsoft data.

The USCI audio and high-speed output routes through the DSMs to the remote sites. The microwave channels are not party-lined; therefore, each channel for each site requires one transmit and receive modem pair. Since the USCI is designed to drive only one 600 $\Omega$  load, the SDA multiplexes the required number of transmit modems for the required number of simulcast sites. For every remote site, there is a corresponding channel bank and Simulcast Serial Adapter (SSA) at the prime site. Each of the SSAs communicates with the DSMs located in the channel banks at the prime site.

At the remote site, the DSM RX modem output routes the audio and data to the base station. (Digital Path systems do not require an RDM since this is clear-only analog.) The delay and amplitude can be set remotely using the SSA and the Prime Optimization Node (PON).

The PTT signal in a Digital Path system originates in the prime site central controller during trunking operation and is sent to the USCI. During failsoft, PTT originates at the USCI and is routed to the M-lead input of the DSM. The E-lead output of the receiving modem is connected to the DSM PTT input and routed to the repeater PTT input. PTT begins at the prime site central controller, is routed from the USCI through the SDA distribution amplifier to the channel bank and DSMs, and on to the repeater.

The receive path of the Digital Path simulcast system begins with the base station repeater. The inbound RF signal from the subscriber radio is received at the repeater and demodulated. The signal is then routed to the transmit line of the DSM for transmission to the prime site. The DSM converts the received audio voltage to digital samples which are applied to the transmit line interface units PCM bus in the channel bank. The transmit line interface unit places the digital samples from the DSM in the channel bank into its assigned time slot in the resulting DS-1 Time Division Multiplex (TDM) digital signal. The DS-1 signal is applied to the input of the digital microwave radio or telco facility. At the prime site, the received DS-1 line interface unit demultiplexes the DS-1 and delivers the appropriate digital signal to the DSM corresponding to the sending DSM at the remote site. The demodulator function of the DSM restores the digital samples to voltages which are then filtered to produce the desired audio signal frequencies. The receive audio signal is routed to the receiving ASTRO-TAC 3000 voting comparator. The comparator selects the best signal based on the highest signal-to-noise ratio. This signal, called the voted audio output, is then sent to the USCI if the system is in failsoft mode, site trunking or stand-alone simulcast. If the simulcast system is part of a SmartZone system, the voted audio is sent up to the master site.

## Failsoft Operation

During failsoft operation, the users are still able to use the voice channels in a conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals this condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeed data. After seeing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC 3000 comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.

## Mixed Mode Using Analog Microwave

The analog microwave mixed-mode simulcast configuration uses signals from the GPS system to synchronize the digital audio from multiple transmitters (see Figure 1-21).

Between the comparator at the prime site and each of the remote site transmitters, there are separate digital and analog links. The digital and analog audio is transported on two separate voice channels in the microwave multiplex system. Analog audio is carried over special wideband analog voice cards. Analog audio is party-lined for the distribution. Digital audio is carried over Single-Sideband (SSB) voice cards using 9600 bps synchronous modems.

If the subscriber signal is analog, the ASTRO-TAC 3000 comparator votes the best quality analog audio and routes it to either the SmartZone AEB switch or back to the simulcast transmitters. If the inbound subscriber signal is digital, the ASTRO-TAC 3000 at the prime site makes a composite best quality digital signal and routes it to either the SmartZone AEB switch or through the USCI, and then back to the simulcast transmitters. Signals that will travel to the AEB are eventually converted to a digital form and combined into a single DS-0 signal by the channel bank ADPCM, SRU and CPU cards. In most SmartZone analog microwave systems, this is done at the SmartZone prime site. This is required so that the AEB can switch both the digital and analog paths in a single DS-0 signal.

Note that the hybrid digital and analog links within a simulcast sub-system are always carried on separate microwave mux voice channels. The hybrid digital and analog links to a SmartZone Prime site must be combined onto a single time slot prior to entering the SmartZone audio switch.

The system commands the station to turn its transmitter on and off by several different means. PTT comes directly from the remote site controller to the station via the TPTT line on the station parallel control cable. The station can also get an EXT-PTT from the wide band modem's E & M lead signalling provided by the prime site central controller. PTT also is signalled via the embedded data that the station receives with the digital voice information. Any of these methods will cause the station transmitter to key.

The station will always key in analog mode if either the TPTT or EXT-PTT are asserted without the presence of ASTRO data on the V.24 port. If ASTRO data is

**Mixed Mode Using ASTRO-TAC 3000**

present on the V.24 port, the station will key in ASTRO mode regardless of the TPTT/EXT-PTT state. If ASTRO data stops suddenly (such as due to a broken data link), the station will stop transmitting even though TPTT/EXT-PTT are still asserted. This is done to prevent the station from erroneously transmitting a silent analog signal when the stations at other simulcast sites are all transmitting the proper digital simulcast signal.

**Digital Operation**

The ASTRO-TAC 3000 comparator has separate transmit and receive paths that are each processed independently. The comparator receive process generates a composite data stream from the best quality signals from the active receivers. Since the same data is often available from many receive sites at the same time, the comparator can often generate a composite signal of higher quality than any of the individual data signals.

For the ASTRO-TAC 3000 transmit process, digital voice signals enter the comparator either from the comparator receive process (system functioning in site-trunking mode) or from another site via the SmartZone audio switch (system functioning in wide-trunking mode). Next, the ASTRO-TAC 3000 comparator embeds the equivalent of analog lowspeed data in the digital voice channel stream. This data comes from the prime site controller to the comparator via the DBB. This embedded data Link Control (LC) data tells the radio that it has reached the correct voice channel. The LC data also facilitates the operation of the dispatcher interrupt and priority monitor features.

The ASTRO-TAC 3000 comparator embeds a digital time stamp in the outbound data stream every 180 msec (see Figure 1-7). Identical time stamps reach each station. This time stamp is then used by the simulcast station to know when to transmit the outbound audio. When the digital voice signals arrive at the simulcast stations, they are held until the time specified by the embedded time stamp. When the proper time arrives, the station sends the signal onto the RF channel.

The GPS receiver/site standard provides the 1 PPS signal that is used to generate the time stamps at the prime site comparator. Another GPS site standard at each of the remote transmitter sites provides a 1 PPS signal that the station uses to determine when to launch the voice information onto the RF channel. The use of this satellite time signal allows the time-frequency reference of the prime site and all the remote sites to be synchronized to within 1  $\mu$ sec.

The digital voice signals are transported between the prime site and remote sites via V.24 links. V.24 is the international equivalent to the EIA-232 specification. These links are synchronous; framing for the bits is provided by separate clock signals for both transmit and receive. The station and comparator also make use of the RTS, CTS, and CD lines.

In the dual path analog microwave system, the V.24 links are typically connected between prime and remote sites using V.3225, 9600 bps modems. Each of these modems is then connected to a normal 4 kHz SSB duplex voice card in the analog microwave mux shelf.

Trunked systems use analog channel equipment for transport of the control channel data. This means that wide-band modems must be used for transport of the trunking control channel signals within a simulcast sub-system.

## Analog Operation

The ASTRO-TAC 3000 comparator used for analog voting also has separate transmit and receive paths. The comparator receive process continuously selects the best signal-to-noise ratio signal available from all unsquelched receivers. This voted audio output is then available to the SmartZone switch or the comparator transmit process.

In the ASTRO-TAC 3000 comparator transmit process, analog voice signals enter the ASTRO-TAC 3000 comparator either directly from the receive process (system in wide- or site-trunking) or from this or another radio site via the SmartZone audio switch (system in wide-trunking). The voted output of the ASTRO-TAC 3000 is passed into the USCI. The USCI is responsible for mixing/gating the repeat audio with the high-speed data from the prime site central controller and for adding lowspeed data. The USCI also generates failsoft data.

From the USCI, audio and high speed data are sent down one path, while lowspeed, failsoft and disconnect data are Frequency Shift Keying (FSK) encoded and sent down the second path. The USCI also generates failsoft data.

The USCI audio and high-speed output goes to a single Starplus transmit-only wideband modem, while the FSK output routes to a single Starplus transmit-only single sideband modem. The audio and high-speed path uses wideband modems because high-speed data requires a bandwidth greater than single sideband modems can provide. Also, the wideband modems have linear phase response which allows for audio phasing by adding linear delay to the audio path.

Both the wideband and SSB multiplex paths are set up in a party-line configuration. This configuration uses a single transmitting modem with multiple receiving modems. This design reduces the total baseband loading by reducing the number of carriers on the microwave baseband.

At the remote site, the receiving multiplex modems provide the audio, high-speed, and FSK inputs to the Remote Delay Module (RDM). The RDM converts the FSK to lowspeed data and provides amplitude and linear delay adjustment. The RDM also provides these same adjustments separately for the audio and high-speed via the Primary Optimization Node (PON). From the PON at the prime site, the installer addresses a specific RDM, enters the amplitude and delay values, and then transmits them over a data link to the remote site. The delay adjustment on the RDM allows all transmitters on a single frequency to transmit all signals in phase within  $\pm 2.6 \mu\text{sec}$ .

Transmitting all signals in phase prevents one form of simulcast distortion. The amplitude adjustment on the RDM allows the audio, high-speed, and low speed data (which includes failsoft and disconnect data) to be adjusted separately in 0.05 dB increments. This causes all transmitters on a single frequency to have the same deviation, which prevents another form of simulcast distortion. This fine-tuning adjustment compensates for level errors that may have occurred at one or more of the many adjustment points in the repeat audio path.

**Mixed Mode Using ASTRO-TAC 3000**

After the delay and amplitude adjustments occur on the RDM, the audio, highspeed and lowspeed data paths are summed in the RDM. The single output of the RDM is connected to the base station. At the base station, the audio/data is fed to the base station's wideband input.

For the analog portion of a mixed-mode channel using analog microwave, the PTT signal originates in the prime site central controller during trunking operation and is sent to the USCI. During failsoft, PTT originates at the USCI and is routed to the M-lead of the transmit only wideband multiplex modem. The E-lead output of the receiving modem is connected to the RDM PTT input and the RDM PTT output is connected to the base station PTT input. PTT is routed from the USCI through the multiplexer, through the RDM and on to the base station.

The receive path begins at the point where the simulcast base station receives RF signals and ends at the prime site's voting comparator. The inbound radio subscriber RF signals are received at the simulcast base station and demodulated. The audio portion of the signal is routed to a Starplus transmit-only SSB modem. The Starplus modem sends the audio signals to a receive-only SSB modem at the prime site. From the multiplex modem, the audio signals are routed to a ASTRO-TAC 3000 voting comparator. The comparator selects the best signal based on the highest signal-to-noise ratio. This signal, called the voted audio output, is then sent to the USCI if the system is in failsoft mode, site trunking or stand-alone simulcast. If the simulcast system is part of a SmartZone system, the voted audio is sent up to the prime site. The trunking control channel is treated and handled as an analog voice channel.

**Failsoft Operation**

During failsoft operation, the users are still able to use the voice channels in conventional repeat mode. In failsoft mode, all site transmitters are turned on. On a digital failsoft channel, the embedded voice channel data is changed to a pattern that signals the failsoft condition to the subscribers. On an analog failsoft channel, the USCI generates failsoft lowspeed data. After sensing that the control channel is gone and not finding any other site control channels, the subscribers lock onto these failsoft voice channels. The selection of analog or digital mode for the failsoft channel is made by the ASTRO-TAC 3000 comparator RSS. The subscriber automatically receives and is then able to respond in the mode signaled by the infrastructure.

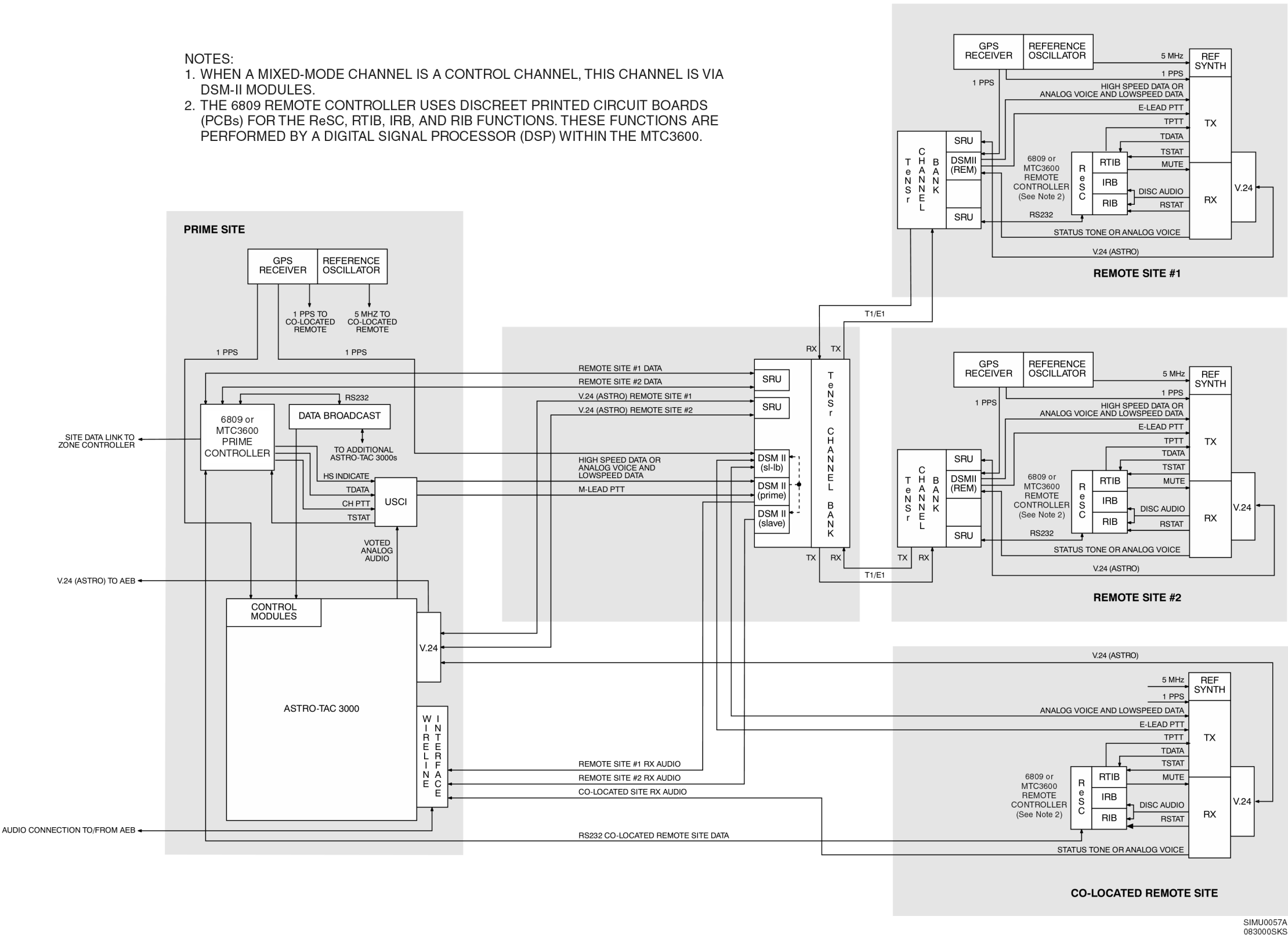


Figure 1-19 Digital Path ASTRO-TAC 3000 Mixed Mode With TeNSr Channel Banks (SmartZone Subsystem)

NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

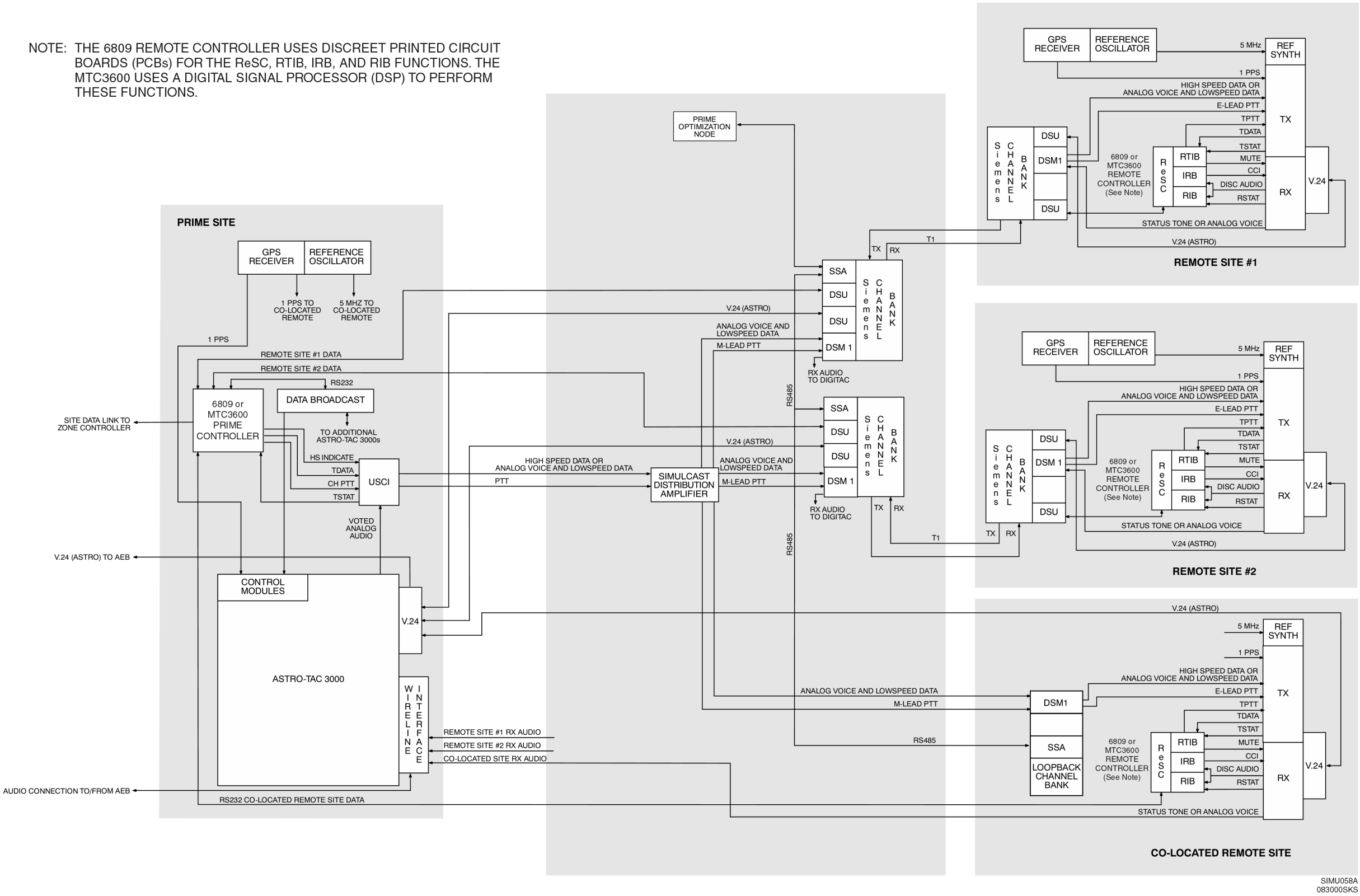


Figure 1-20 Digital Path ASTRO-TAC 3000 Mixed Mode With Siemens Channel Banks (SmartZone Subsystem)



NOTE: THE 6809 REMOTE CONTROLLER USES DISCREET PRINTED CIRCUIT BOARDS (PCBs) FOR THE ReSC, RTIB, IRB, AND RIB FUNCTIONS. THE MTC3600 USES A DIGITAL SIGNAL PROCESSOR (DSP) TO PERFORM THESE FUNCTIONS.

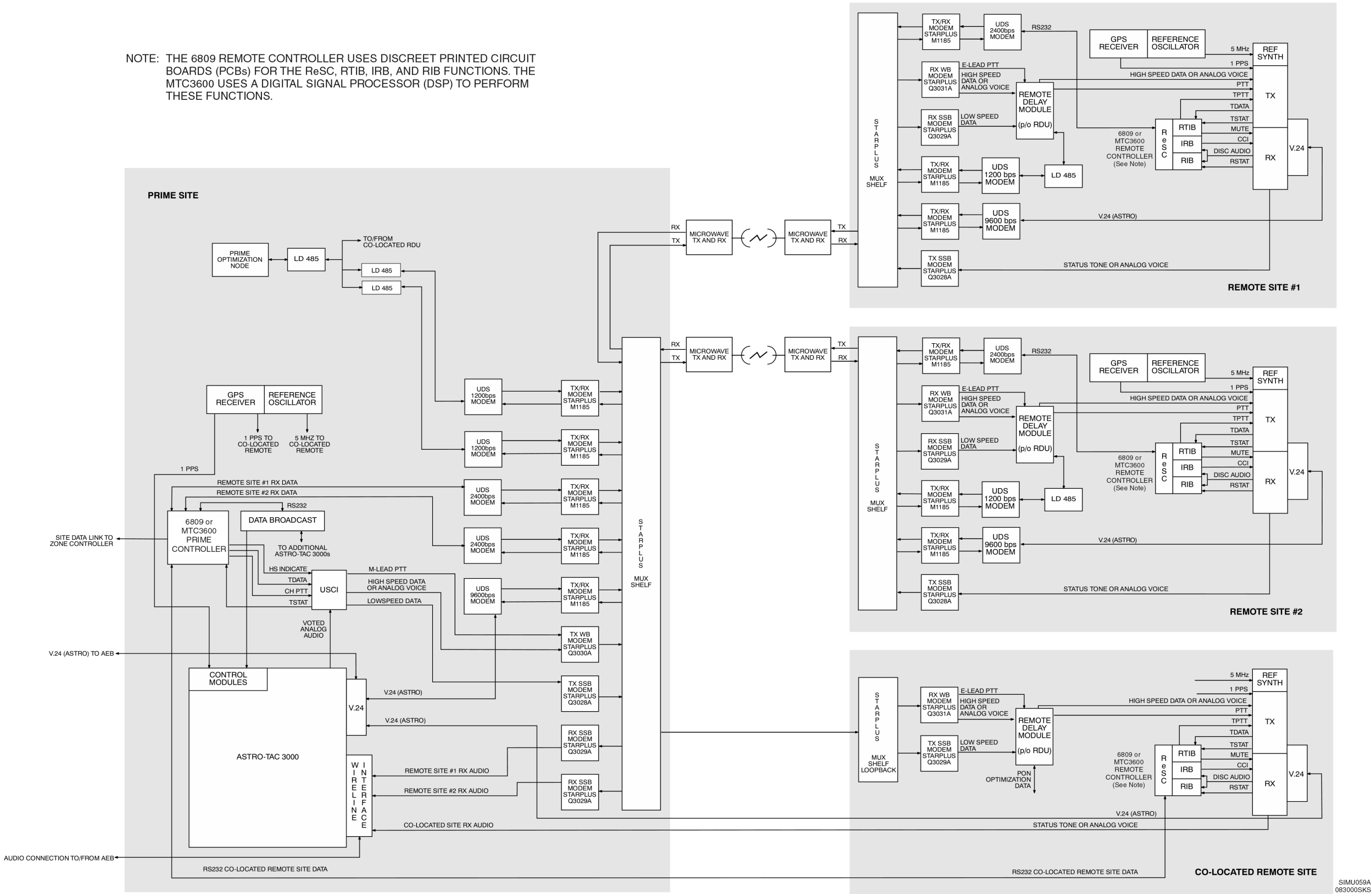


Figure 1-21 Digital Path ASTRO-TAC 3000 Mixed Mode With Analog Microwave (SmartZone Subsystem)

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# ***Rack Installation***

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## **Overview**

This chapter provides instructions for installing the equipment racks. The following topics are included in this chapter:

Section	Page	Description
Unpacking And Inspection	2-2	Provides general unpacking and inspection items.
Rack Installation	2-3	Provides general information on installing system equipment racks.

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## Unpacking And Inspection

<b>NOTE</b>
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Thoroughly inspect the equipment as soon as possible after delivery. If any part of the equipment has been damaged in transit, immediately report the extent of the damage to the transportation company.

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<b>NOTE</b>
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Before discarding packing materials, make certain any hardware or informational material is retained.

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<b>NOTE</b>
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Throughout this manual, Motorola accessories are identified by Motorola kit number (part number begins with a "B") as labeled on packaging. Actual part number on assembly may be different.

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The equipment is shipped by electronic van.

Unpack the equipment as follows:

1. Move the carton as close to the installation location as convenient.
2. Carefully cut the four straps securing the carton to the wooden skid.
3. Remove the carton from the rack and skid.
4. Carefully hoist the rack into the upright position.
5. Carefully cut the strap securing the inserts. Remove the top insert.
6. Remove the four bolts securing the rack to the bottom insert.

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## Rack Installation

### Planning the Installation

Read the entire procedure before beginning the installation. It is essential that you carefully plan the installation before actual work is started, because properly installed equipment is essential to obtaining optimum system performance. Consider the following:

- ☐ The location of the equipment in relation to power and communication lines.
- ☐ Convenient access for servicing the equipment.
- ☐ The size of the racks.
- ☐ Ensuring that all tools, equipment, and facilities are available before beginning the installation.

### Rack Mounting

Typical installations use a standard 90 inch equipment rack.

Use 0.5 inch diameter bolts to anchor the rack to the floor. Recommended anchoring methods for different floor types are provided in Table 2-1.

Table 2-1 **Recommended Anchoring Methods**

Floor type	Methods
Steel	Drill and tap; use 0.5" diameter bolts.
Wood	Floor must be 2.54 cm (1 in.) or thicker. Use 0.5" diameter lag bolts.
Concrete	Use screw anchors of at least 0.5" diameter.

### Ground Connections

#### **! WARNING !**

Proper site grounding and lightning protection are vitally important considerations. Failure to provide proper lightning protection could result in injury to personnel and catastrophic damage to the equipment.

Grounding shall be performed in accordance with the Motorola reference publication, *Quality Standards - FNE Installations R56*, see "Recommended Reference Publications," page 1-2.

## Rack Installation

Each equipment rack is equipped with two ground lugs, located at the top and at the base of the frame. Use one of these lugs to connect the rack to the site ground point.

### ! WARNING !

All telephone lines, antenna cables, and AC or DC power cabling must be properly grounded and lightning protected by following the rules and guidelines provided in the Quality Standards reference. Failure to provide proper grounding and lightning protection could result in injury to personnel and catastrophic damage to the equipment.

## Rack Mounting Requirements

Space and weight requirements for the equipment are provided in Table 2-2.

Table 2-2 **Rack Space/Weight Requirements**

Unit	Weight	Height	Width	Depth
Premisys TeNSr 800 Channel Bank	11 lb	9 in.	17 in.	15.38 in.
Premisys TeNSr 600 Channel Bank	8.5 lb	9 in.	17 in.	9.13 in.
USCI/CSCI Card Cage with power supply	11.8 lb	8.75 in.		9.0 in.

## Power Requirements

### Premisys TeNSr Channel Banks

Refer to the *TeNSr Manual* for power requirements, see “Recommended Reference Publications,” page 1-2.

### USCI/CSCI

The USCI and CSCI use a modular power supply that slides into the card cage. Each card cage has a dedicated power supply that can drive up to eight modules.

# ***Conventional Simulcast Controller Interface***

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## **Overview**

This chapter provides instructions for installing and setting up the CSCI. The following topics are included in this chapter:

Section	Page	Description
Description	3-2	Describes the role of the CSCI in a conventional simulcast system.
Theory of Operation	3-3	Describes the CSCI circuitry.
CSCI User Interface Switches, Controls, And Connectors	3-8	Lists CSCI connectors, interfaces, and switch settings for typical configuration setups.
Installing The CSCI	3-15	Provides instructions for installing the CSCI.
Troubleshooting	3-19	Provides CSCI field troubleshooting information and corrective action.

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**Description**

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**Description**

In a simulcast radio system, the Conventional Simulcast Controller Interface (CSCI) generates Private Line (PL) or Digital Private Line (DPL) tones and sums it with the transmit audio. In addition to the summing, the CSCI pre-emphasizes and limits the transmit audio signal.

In two-level and four-level systems, each base station channel requires a CSCI module, part number QTN6066B. Conventional Simulcast Card Cage, part number Q3367A, houses up to eight CSCI modules and a single system can have a maximum of four Conventional Simulcast Card Cages. Each individual card cage is powered by 13.8 VDC power supply, part number TPN1153A. A 25-pair telco-type cable connects four CSCI modules to one punchblock (for distribution to the microwave or T1/E1 lines).



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## Theory of Operation

Refer to the schematic diagrams for the following explanations.

### Four-Level Systems

In four-level SECURENET® simulcast systems, coded audio is processed through the CSCI modules coded audio path, and is keyed by the CODED INDICATE IN logic signal from the comparator. The actual four-level conversion of the coded audio occurs in the DSM-II at the transmit sites.

### Transmit Audio Circuitry

The transmit audio input to the CSCI module is compressed (869 MHz systems), pre-emphasized, clipped, summed with a PL or DPL tone (which is generated on board), and low-pass filtered for transmission. The clear transmit audio circuitry (PL/DPL gate muted) has the following characteristics:

- ☐ Input Level (1 kHz test tone):
  - ☐ -10 dBm with no compression
  - ☐ -8.1 dBm with compression
- ☐ Input Termination: 600 $\Omega$  balanced
- ☐ Pre-emphasis: 6 dB per octave from 300 Hz to 3 kHz
- ☐ Clipping: limits output to 80% deviation at transmitter for clear audio
- ☐ Low pass filter: corner frequency = 8 kHz
- ☐ Maximum output level: 700 mVp-p typical; 740 mVp-p maximum
- ☐ Output termination: 600 $\Omega$  balanced
- ☐ Signal-to-noise: 50 dB minimum for clear audio (compression circuitry off and PL/DPL gate muted)
- ☐ Frequency Response: +1 to -3 dB from 300 Hz to 3 kHz for 6 dB per octave pre-emphasis with input level of -20 dBm
- ☐ Distortion: less than 1% for 1 kHz test tone input of -15 dBm

### Differential Input Stage

The differential transmit audio enters the CSCI on pins 2 and 3 of edge connector J1A. Each input leg is applied across a 600 $\Omega$  impedance consisting of an operational amplifier (op amp) buffer for each line input whose output feeds the input of an op amp differential stage. The differential scheme is used to enhance noise rejection through the high common-mode rejection inherent with differential signals. This stage also transforms the differential signal into a single-ended signal (U100-1) for driving the next stage. Switch S100-1 allows

## Theory of Operation

muting of the audio path through the CSCI during system setup and optimization procedures.

### Coded Indicate

The comparator tells the CSCI when a coded call is in progress by pulling J1A -15 low. U107-4 pulls U103-11 low to select the coded audio path when CODED INDICATE IN is low. When U107-4 is low, it also pulls U103-10 low through CR104. With U103-10 low, the compression circuitry is disabled; this eliminates any noise generated by the compression circuitry from being summed at the output during a coded operation. The U107-4 output also pulls U101-6 low through CR117, disabling the output of the PL/DPL generating circuitry during coded operation.

During a coded call U107-10 is low, which turns on opto-isolator U110. CODED INDICATE OUT is a floating output, so it can be connected to -24 VDC or -48 VDC microwave equipment. CODED INDICATE OUT(+) at J1A-7 connects to the more-positive potential on the M-lead, and CODED INDICATE(-) at J1A-6 connects to the more-negative potential on the M-lead. Since microwave equipment operates on negative DC voltages, CODED INDICATE OUT(-) typically connects to a fixed negative voltage, and CODED INDICATE OUT(+) connects to the switchable M-lead input.

### Clear/Coded Audio Path Switching

The output of the audio path MUX gate, U101-10, is applied to the clear coded analog MUX gate U103-14. This MUX gate passes the audio to the clear audio path when CODED INDICATE IN at J1A-15 is not held low. After passing through MUX gate U103-14, the demultiplexed clear audio at U103-13 is selectively applied to either the compression stage or a simple gain stage. The coded audio signal at U103-12 is applied through a saturation gain stage which squares the DVP data. The two paths are combined through summing amplifier U105-9.

### Compression

The audio signal branches into two paths at U102-14. For systems requiring no compression, the audio passes through a simple gain amplifier ( $A = 1.545$ ). For systems requiring compression, such as with 896 MHz trunking, the audio is compressed by 2:1 compressor U106. The peak-to-peak voltage at U103-1 ( $V_o$ ) is expressed as follows:

$$V_o = \sqrt{2.19(V_i)}$$

$$(V_i = V_{p-p} \text{ @ U102-4; } f_o, 0\text{dB})$$

Compressed and non-compressed audio are combined at MUX gate U103. Compressed audio is present at U103-15 only if switch S101-2 is open and CODED INDICATE IN is high (clear audio). Non-compressed audio is present at U103-15 if switch S101-2 is closed or CODED INDICATE IN is low. (For encrypted voice, the compressor is turned off to prevent high levels of noise at the output of the compressor when no signal is present at its input.)

## Pre-emphasis

Once compressed or non-compressed audio is chosen at U103-15, the clear audio is passed through the pre-emphasis filter. The pre-emphasis filter frequency response increases 6 dB per octave from 300 Hz to 3 kHz. The response levels off at 12 kHz and starts to fall off at 6 dB per octave at 15 kHz. The gain of the pre-emphasis stage at 1 kHz is 1.82.

## Slew Rate Limiter

Severely filtered square waves can contain overshoot as great as 20%. To minimize overshoot, the CSCI employs a slew rate limiter to prevent the formation of square waves when audio is in hard clip. The pre-emphasized audio from U100-14 is lowpass-filtered and applied to U104-13. As long as the amplitude of the waveform at U104-13 is less than the limiter's threshold, diodes CR109-1 and CR109-2 are off; in this state U104-12, U104-13, and U104-7 constitute a unity gain voltage follower. When the input signal becomes large enough, the current through R133 turns on either CR109-1 or CR109-2. Under this condition, the voltage follower becomes a comparator driving an integrator, which limits slew rate. The input voltage at U104-13 is compared to the output voltage of the integrator, U104-7, and forces the voltage at U104-7 to be equal to the voltage at U104-13. When the signal becomes large enough to be limited, the voltage at U104-14 is rail to rail, and the diodes clamp that voltage across R135. R135 and C112 form an integrator with U104-7 slewing at the rate of  $V_{diode}/(R176 \cdot C112)$  V/s. The slew rate for this circuit is 55.6 V/ms; at the audio output, the slew rate is 3.6 V/s.

## Amplitude Limiter

From the slew rate limiter, the transmit audio signal is applied to the amplitude limiter. The supply voltage at U104-4 determines the maximum voltage output at U104-7. If the output voltage of the CSCI module is too high, the power supply voltage can be reduced. Refer to power supply schematic diagram, 6881039E36, for circuit information (see "Recommended Reference Publications," page 1-2).

CR129 is placed between the 13.8V power supply rail and U104-4 to provide a symmetrical output at U104-1. All of the circuitry after the limiter is based on the assumption that the limiter output at U104-1 is 11.15Vp-p (which is the maximum output level), and the level at the audio outputs is 705 mVp-p.

In cases where overdeviation occurs, the limiter output amplitude is decreased accordingly using potentiometer R112, which controls the power supply voltage applied to the limiter. (Since the limiter operates in full saturation, adjusting its output amplitude is accomplished by merely changing the power supply voltage applied to the limiter.) R112 cannot change the maximum peak-to-peak output of the CSCI module. The gain for the limiter stage in the linear region is 3.39.

**Theory of Operation****Summing Amplifier**

U105-9 sums the three CSCI audio inputs (clear audio from U104-1, coded audio from U104-8, and PL/DPL modulation from U101-9). Clear audio can be sent from the module with or without the PL/DPL component.

The summing amplifier provides four different gains as follows:

- ☐ 0.063 for clear audio
- ☐ 0.048 for coded audio
- ☐ Adjustable gain for PL/DPL from 0.038 to 0.19 (dependent on R141 setting)
- ☐ Unity gain for the audio input at high impedance input jack J2 (which receives its signal from U102-8)

**Splatter Filter**

Following the summing amplifier, the audio signal passes through splatter filter U105-1, which attenuates the higher harmonics of high-speed data and coded audio. The filter has a 3-pole Bessel response with a cutoff frequency of approximately 8 kHz. The Bessel response is used due to its linear phase response.

**Differential Line Driver**

The output of the splatter filter feeds the audio output differential line driver. Because there is an odd number of inverting stages in the audio path, the noninverting driver (U105-14) provides the AUDIO OUT(–) signal and the inverting driver (U105-7) provides the AUDIO OUT(+) signal. The audio outputs are terminated into 600 $\Omega$  loads connected across J1A-4 and J1A-5. With the audio output level adjusted for -10 dBm, the signal at J1A-5 adds to the inverted signal at J1A-4 to yield a 693 mVp-p signal.

**PL/DPL Generation Circuitry**

The CSCI module contains ICs and switch/resistor matrixes that provide the generation and selection of supported PL/DPL codes.

U112 is a programmable Tone Encoder/Decoder that generates the PL tones. The decoding capabilities of the IC are not used. U112 receives 9.1 VDC operating voltage from regulator stage Q100, which receives its power from the 13.8 VDC rail. The output of U112 output is applied to an amplifier and then a low-pass filter stage. The output from the filter stage is then gated through U103 and U101 to the summing amplifier stage. The 3dB corner of the low-pass filter is approximately 250 Hz. For PL frequencies from 67 to 79.7 Hz, the filter 3dB corner is moved lower by capacitor C170 being connected across capacitor C159. Capacitor C170 is connected across C159 when switch S102-10 is closed. The C159/C170 parallel combination filter cutoff frequency improves the distortion characteristics at the lower PL frequencies. The PL frequencies are set using switch S103, which develops codes used by tone encoder U112. Refer to the PL Frequency Select Switches section, Table 3-4, for PL code settings.

Digital Sequence Detector U114 serves as a DPL encoder IC. Decoding capabilities of the IC are not used. U114 receives 6.2 VDC operating voltage from a voltage regulator consisting of Zener diode VR101 and R178, which is powered from the 13.8 VDC rail. The DPL IC output at U114-5 splits into two paths. The first path is applied directly through buffer amplifier Q105. The second path is applied to inverter stage Q103. The inverted output is selected when switch S101-3 is closed.

The buffered output at Q105 is applied through a low-pass filter that has a corner during PTT of approximately 85 Hz for filtering the digital word. When the PTT is released, Q102 turns off. With Q102 off, the filter corner is set to 140 Hz. The 140 Hz corner filters the 134 Hz turn-off-code. The signal is then applied to Q106 and Q107, which provides a high-input impedance, high-gain output stage.

Switch S102 sets the DPL codes developed by DPL IC U114. Refer to the DPL Code Selection Switch section, Table 3-3, for DPL code settings.

### PTT / Reverse Burst / Turn-Off-Code Operation

The PTT logic (active low) enters the module at J1A-10. The output of U108-6 turns on Q101, which is connected to U112-26 and U108-9. With PTT operation enabled, the logic passes through U101 and then to U107-11. This, in turn, activates the red PTT LED and U107-14. With U107-14 on, opto-isolator U111 also turns on. PTT opto-isolator closure outputs U111-4 and 5 are then fed to card edge connectors 11 and 13.

When the PTT input to the CSCI module is released, U112 internal circuitry senses the PTT release and initiates the “Reverse Burst” function that extends the period that the PTT signal at U112-26 remains low. During the same period, U112 internal circuitry also changes the phase of the PL tone being generated and sent out of the CSCI module. The reverse burst time period is determined by an R-C circuit connected to U112. The reverse burst timing circuitry consists of R188, VR100, CR125, CR126, C134, and R191. The values of R191 and C134 are selected to generate the PTT extension for approximately 150 ms.

The DPL stages operate in response to PTT logic as follows:

- ❑ When PTT logic at J1A-10 switches low, U108-4 switches high. The signal at U108-4 is applied to U114-9 which, when switched high, starts the DPL word output. At the same time, Q102 turns on and modifies the low-pass filter response during the generation of the DPL word.
- ❑ When the PTT is released U114-9 returns low, which switches the output from DPL word to Turn-Off-Code. Also, Q102 turns off, returning the filter to the proper response curve for processing the Turn-Off-Code.

The PL operation uses the same Reverse Burst circuitry as the DPL operation to extend the PTT while the Turn-Off-Code is being generated.

## CSCI User Interface Switches, Controls, And Connectors

The switches, controls, and connectors discussed in the following section aid in optimization and troubleshooting, and are used to configure the CSCI for various modes.

### Switch S100 Test Modes

Switch S100 inhibits module input signals to allow system troubleshooting and optimization. This four-section Dual In-Line Packet (DIP) switch is located on the front of the CSCI module. Closing any one of the four sections (S100-1 through S100-4) causes the green power LED to blink on and off, indicating some of the audio or logic circuitry has been disabled for testing.

#### NOTE

When any of switch S100 sections are closed (power LED blinks), calls through the module will not be processed.

The individual sections of the switch allow selective signal disabling. For example, when the PTT Disable section is closed, a PTT will not be generated. However, the other module outputs (such as CODED INDICATE OUT and its coded audio) will remain enabled.

Table 3-1 lists the various test settings for switch S100, as well as the functions of each setting.

Table 3-1 **Switch S100 Test Settings and Functions**

Switch S100 Section/ Setting	Function
Mute Audio Path (S100-1 closed)	Applies logic ground to U101-12, removing the audio path from U100-1 to U103-14. Used during optimization of the PL/DPL levels when audio may be present at the module audio input and needs to be muted.
PL/DPL Mute (S100-2 closed)	Applies logic ground to U101-6 through CR116, thereby opening the path from the PL/DPL generating circuitry to the summing amplifier input. Used during clear audio level optimization and level setting to remove the PL/DPL modulation component during level setting.
Coded Indicate Disable (S100-3 closed)	Disables the CODED INDICATE IN logic path. In turn, CODED INDICATE OUT at U110-6, 7 is inhibited. (If there is coded audio at the module input pins it will be processed through the clear path.) Used for inhibiting CODED INDICATE IN during testing.
PTT Disable (S100-4 closed)	Applies logic ground to U101-5, which inhibits the output of PTT logic at card edge connectors 11 and 13 (U111-4, 5). Used for disabling PTT output during system optimization.

## Switch S101 Function Settings

DIP switch S101 selects several CSCI functions, listed in Table 3-1.

Table 3-2 **Switch S101 Functions**

Switch S101 Section	Function
DPL Normal/Invert (S101-1)	When set to open position, non-inverted DPL signal is applied through amplifier and filter stages. When set to closed position, DPL signal is inverted through Q103 and then applied through amplifier and filter stages.
Compression Enable/ Disable (S101-2)	When set to open position, compression is enabled. (Open position is normal position for 896 MHz systems.) When set to closed position, compression is disabled.
PL/DPL Select (S101-3)	When set to open position, PL tones are applied through summing amplifier stage. When set to closed position, DPL tones are applied through summing amplifier stage.
(S100-4)	Not used.

## Potentiometers R112 and R141

CSCI front panel potentiometer R112 sets the audio output level of clear audio. This adjustment does not affect the clipping level. R112 allows an approximate  $\pm 3$  dB gain adjustment of the non-compressed audio path.

CSCI front panel potentiometer R141 adjusts the gain in the PL/DPL audio path to the summing amplifier. The range of adjustment is approximately 10% to 20% of system deviation (500 Hz to 1000 Hz of modulation depth).

## Phone Jacks J2 and J3

J2 is a high impedance, transmit path optimization input. As an aid in system optimization, it allows the injection of test tones into the transmit path prior to the summing circuitry of the CSCI.

J3 is a bridged monitor for the audio input signal. When the jack is used to monitor the incoming signals, the audio level on the line is not affected.

### NOTE

If J3 is used as an input, remove any connection to the audio inputs on the punchblock. This prevents double termination of the input.

## CSCI User Interface Switches, Controls, And Connectors

## DPL Code Selection Switch

Table 3-3 lists the settings for switch sections 1 through 9 on switch S102 that select the DPL codes.

## NOTE

In Table 3-3, “0” denotes switch set to OPEN position, “1” denotes switch set to CLOSED position.

Table 3-3 DPL Code Select Matrix

DPL Code	Switch S102 Section								
	1	2	3	4	5	6	7	8	9
023	1	1	1	1	0	1	1	0	0
025	1	1	1	1	0	1	0	1	0
026	1	1	1	1	0	1	0	0	1
031	1	1	1	1	0	0	1	1	0
032	1	1	1	1	0	0	1	0	1
043	1	1	1	0	1	1	1	0	0
047	1	1	1	0	1	1	0	0	0
051	1	1	1	0	1	0	1	1	0
054	1	1	1	0	1	0	0	1	1
065	1	1	1	0	0	1	0	1	0
071	1	1	1	0	0	0	1	1	0
072	1	1	1	0	0	0	1	0	1
073	1	1	1	0	0	0	1	0	0
074	1	1	1	0	0	0	0	1	1
114	1	1	0	1	1	0	0	1	1
115	1	1	0	1	1	0	0	1	0
116	1	1	0	1	1	0	0	0	1
125	1	1	0	1	0	1	0	1	0
131	1	1	0	1	0	0	1	1	0
132	1	1	0	1	0	0	1	0	1
134	1	1	0	1	0	0	0	1	1
143	1	1	0	0	1	1	1	0	0
152	1	1	0	0	1	0	1	0	1
155	1	1	0	0	1	0	0	1	0
156	1	1	0	0	1	0	0	0	1
162	1	1	0	0	0	1	1	0	1



## CSCI User Interface Switches, Controls, And Connectors

Table 3-3 DPL Code Select Matrix (continued)

DPL Code	Switch S102 Section								
	1	2	3	4	5	6	7	8	9
165	1	1	0	0	0	1	0	1	0
172	1	1	0	0	0	0	1	0	1
174	1	1	0	0	0	0	0	1	1
205	1	0	1	1	1	1	0	1	0
223	1	0	1	1	0	1	1	0	0
226	1	0	1	1	0	1	0	0	1
243	1	0	1	0	1	1	1	0	0
244	1	0	1	0	1	1	0	1	1
245	1	0	1	0	1	1	0	1	0
251	1	0	1	0	1	0	1	1	0
261	1	0	1	0	0	1	1	1	0
263	1	0	1	0	0	1	1	0	0
265	1	0	1	0	0	1	0	1	0
271	1	0	1	0	0	0	1	1	0
306	1	0	0	1	1	1	0	0	1
311	1	0	0	1	1	0	1	1	0
315	1	0	0	1	1	0	0	1	0
331	1	0	0	1	0	0	1	1	0
343	1	0	0	0	1	1	1	0	0
346	1	0	0	0	1	1	0	0	1
351	1	0	0	0	1	0	1	1	0
364	1	0	0	0	0	1	0	1	1
365	1	0	0	0	0	1	0	1	0
371	1	0	0	0	0	0	1	1	0
411	0	1	1	1	1	0	1	1	0
412	0	1	1	1	1	0	1	0	1
413	0	1	1	1	1	0	1	0	0
423	0	1	1	1	0	1	1	0	0
431	0	1	1	1	0	0	1	1	0
432	0	1	1	1	0	0	1	0	1
445	0	1	1	0	1	1	0	1	0
464	0	1	1	0	0	1	0	1	1
465	0	1	1	0	0	1	0	1	0
466	0	1	1	0	0	1	0	0	1
503	0	1	0	1	1	1	1	0	0
506	0	1	0	1	1	1	0	0	1
516	0	1	0	1	1	0	0	0	1

**CSCI User Interface Switches, Controls, And Connectors***Table 3-3 DPL Code Select Matrix (continued)*

<b>DPL Code</b>	<b>Switch S102 Section</b>								
	<b>1</b>	<b>2</b>	<b>3</b>	<b>4</b>	<b>5</b>	<b>6</b>	<b>7</b>	<b>8</b>	<b>9</b>
532	0	1	0	1	0	0	1	0	1
546	0	1	0	0	1	1	0	0	1
565	0	1	0	0	0	1	0	1	0
606	0	0	1	1	1	1	0	0	1
612	0	0	1	1	1	0	1	0	1
624	0	0	1	1	0	1	0	1	1
627	0	0	1	1	0	1	0	0	0
631	0	0	1	1	0	0	1	1	0
632	0	0	1	1	0	0	1	0	1
654	0	0	1	0	1	0	0	1	1
662	0	0	1	0	0	1	1	0	1
664	0	0	1	0	0	1	0	1	1
703	0	0	0	1	1	1	1	0	0
712	0	0	0	1	1	0	1	0	1
723	0	0	0	1	0	1	1	0	0
731	0	0	0	1	0	0	1	1	0
732	0	0	0	1	0	0	1	0	1
734	0	0	0	1	0	0	0	1	1
743	0	0	0	0	1	1	1	0	0
754	0	0	0	0	1	0	0	1	1

**PL Frequency Select Switches**

Table 3-4 lists the settings for switch sections 1 through 10 on switch S103 and switch section 10 on switch S102 that select the PL code/frequencies.

**NOTE**

In Table 3-4, “0” denotes switch set to OPEN position, “1” denotes switch set to CLOSED position.

In Table 3-4, asterisk (\*) denotes PL frequency/code not currently supported.

## CSCI User Interface Switches, Controls, And Connectors

Table 3-4 PL Frequency Select Matrix

PL Code	Freq. (Hz)	Switch S103 Section										Switch S102 Section 10
		1	2	3	4	5	6	7	8	9	10	
XZ	67.0	0	1	0	1	0	1	1	0	0	0	1
WZ	69.3	1	1	1	0	1	0	1	1	1	1	1
XA	71.9	1	0	1	0	1	1	1	1	0	1	1
WA	74.4	1	0	0	0	0	1	1	1	1	1	1
XB	77.0	1	0	0	0	0	0	1	1	0	1	1
WB	79.7	1	1	1	0	0	1	1	0	0	1	1
YZ	82.5	1	1	0	0	0	0	1	0	1	1	0
YA	85.4	1	0	0	0	1	0	1	0	0	1	0
YB	88.5	0	1	1	0	1	0	1	1	0	1	0
ZZ	91.5	0	0	0	0	0	1	1	1	1	1	0
ZA	94.8	0	1	1	0	0	1	1	0	1	1	0
ZB	97.4	0	1	1	0	0	0	1	0	0	1	0
1Z	100.0	0	0	0	0	0	1	1	0	1	1	0
1A	103.5	1	1	1	0	1	0	1	1	1	0	0
1B	107.2	1	1	0	0	1	1	1	1	1	0	0
2Z	110.9	1	0	0	0	1	0	1	1	0	0	0
2A	114.8	1	0	1	0	1	1	1	0	0	0	0
2B	118.8	1	1	0	0	0	0	1	0	0	0	0
3Z	123.0	0	1	1	0	0	0	1	1	0	0	0
3A	127.3	0	1	0	0	0	0	1	1	0	0	0
3B	131.8	0	0	1	0	1	1	1	0	0	0	0
4Z	136.5	0	0	0	0	1	0	1	0	0	0	0
4A	141.3	1	1	0	1	1	1	0	1	1	1	0
4B	146.2	1	1	1	1	1	0	0	0	1	1	0
5Z	151.4	1	0	0	1	1	0	0	0	1	1	0
5A	156.7	0	1	1	1	1	0	0	1	0	1	0
5B	162.2	0	0	0	1	1	0	0	1	0	1	0
6Z	167.9	0	0	0	1	1	1	0	0	1	1	0
6A	173.8	1	1	1	1	1	0	0	1	0	0	0
6B	179.9	1	1	1	1	0	1	0	0	1	0	0
7Z	186.2	1	1	0	1	1	1	0	0	0	0	0
7A	192.8	0	1	0	1	0	1	0	1	1	0	0
M1	*203.5	0	1	0	1	0	1	0	0	0	0	0
8Z	*206.5	1	0	1	0	1	1	0	1	1	1	0
M2	*210.7	1	0	0	0	1	1	0	1	1	1	0

**CSCI User Interface Switches, Controls, And Connectors***Table 3-4 PL Frequency Select Matrix (continued)*

PL Code	Freq. (Hz)	Switch S103 Section										Switch S102 Section 10
		1	2	3	4	5	6	7	8	9	10	
M3	*218.1	1	1	0	0	1	1	0	0	1	1	0
M4	*225.7	0	0	1	0	1	1	0	1	0	1	0
9Z	*229.1	0	1	0	0	0	1	0	1	0	1	0
M5	*233.6	0	1	1	0	1	0	0	0	0	1	0
M6	*241.8	1	1	1	0	0	1	0	1	0	0	0
M7	*250.3	1	1	1	0	0	1	0	0	0	0	0

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## Installing The CSCI

The following sections describe how to install the CSCI and associated items.

### Card Cage Installation

Install CSCI card cage(s) as follows:

<b>CAUTION</b>
----------------

System modules contain Electrostatic Discharge Sensitive (ESDS) components. Use ESDS handling precautions, including conductive wrist strap, ESDS-approved work surface, and antistatic bags, when removing or installing module circuit cards. Failure to adhere to ESDS practices may result in immediate or latent equipment failure.

1. Make sure all system power is turned off.

<b>NOTE</b>
-------------

If FRED upgrade is anticipated, leave rack space for the Prime Site FRED card cages. The FRED card cages have the same dimensions as the CSCI card cage. One Prime Site FRED accommodates eight SECURENET channels.

2. Place the CSCI card cage in rack mounting position and secure using four screws, lockwashers, and nuts (supplied). Repeat for each CSCI card cage.
3. Perform interconnection between CSCI card cage, SCI backplane, and PS-FRED (if installed).
4. Proceed to “CSCI Power Supply Connections and Adjustment.”

### CSCI Power Supply Connections and Adjustment

The CSCI uses a modular power supply that slides into the CSCI card cage. Each card cage has a dedicated supply that powers up to eight CSCI modules. Install power supply(s) and perform interconnect as follows:

1. From the back of the rack, carefully route CSCI power cord TLN5960A between the card cage backplane and the bottom of the CSCI card cage.

**Installing The CSCI**

2. From the front of the CSCI card cage, connect the power supply cord to the mating receptacle on rear of the module.
3. From the front of the CSCI card cage, insert the power supply module into its far right slot of the CSCI card cage.
4. While inserting the power supply module, carefully route the four power supply module wires out the back of the CSCI card cage.
5. From the front, tighten the screw which secures power supply module to card cage.
6. From the back of the CSCI card cage, connect the power supply module output cable to card cage backplane terminal block connectors as follows:

<b>Power Supply Module Cable:</b>	<b>Backplane Terminal:</b>
Brown-Red (Regulated A+)	A+ terminal
Yellow-Brown (Power Alert)	ALERT terminal
Black (ground)	GND terminal
Red-Yellow (Switched A+)	SPARE terminal

7. Route the output cable along the side of the rack. Secure cable to rack using cable ties.

<b>NOTE</b>
-------------

The second set of terminals and diodes, CR1 and CR2, on the CSCI main board are used to connect a second power supply for redundancy.

---

Second power supply connects to AUX A+ and GROUND screw terminals on card cage motherboard.

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8. Apply power to the card cage.
9. Connect DC voltmeter across GND and A+ terminals on backplane.
10. Measure A+ voltage and proceed as follows:
  - 10.1 If the reading is 13.8V, proceed to step 11.
  - 10.2 If the reading is not 13.8V, adjust R107 on power supply module. Proceed to step 11.
11. Remove power from the card cage.
12. Proceed to "CSCI Module Installation."

## CSCI Module Installation

### NOTE

The PS-FRED channels must connect to the same CSCI channel (for example, PSFRED channel 1 must connect to CSCI channel 1).

1. On CSCI module, set all stations of switch S100 to the open position.
2. Depending upon configuration required, set switch S101 on CSCI as follows:

Switch S101 Section	Function
DPL Normal/Invert (S101-1)	When set to open position, non-inverted DPL signal is applied through amplifier and filter stages.  When set to closed position, DPL signal is inverted through Q103 and then applied through amplifier and filter stages.
Compression Enable/ Disable (S101-2)	When set to open position, compression is enabled. (Open position is normal position for 896 MHz systems.) When set to closed position, compression is disabled.
PL/DPL Select (S101-3)	When set to open position, PL tones are applied through summing amplifier stage. When set to closed position, DPL tones are applied through summing amplifier stage.
(S100-4)	Not used.

3. Starting with the left slot, insert the Channel 1 CSCI module in the card cage.

### NOTE

Each CSCI card cage slot corresponds to a dedicated channel from the central controller. If a channel is not used, skip the slot.

4. Repeat steps 1 through 3 for each module of each card cage.
5. Proceed to the following “Cabling Installation” section.

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Installing The CSCI

## Cabling Installation

Perform CSCI cabling as described in the following sections.

### **CSCI-to-Punchblock Cabling (from comparator/ Spectra-TAC™)**

Perform CSCI-to-punchblock cabling as follows:

1. From the front of the CSCI, connect the 25-pair cables to J1 and J2 on the CSCI interface board.
2. Secure the cables to J1 and J2 using the velcro straps.
3. Route the cable to the punchblocks.
4. Connect the cable to the punchblock.
5. Connect the punchblock terminal clips to the appropriate connectors.
6. Set levels (as part of optimization), described in the applicable manual, *Dual Path Simulcast Trunked System Optimization* or *Digital Path Trunked Simulcast System*. See “Recommended Reference Publications,” page 1-2.



## Troubleshooting

Due to the advanced technology and manufacturing process of Motorola trunked system equipment, technicians should not attempt to repair any boards in the field. Motorola recommends that only authorized Motorola service depots make module repairs. Before sending a module for service, check to see if one or more of the following potential problems is affecting the proper operation of the CSCI. If any of the tests indicate a failure, include this information when you send the module to a Motorola service depot.

The following procedures establish the location of the malfunction and determine if the CSCI is at fault. To verify module functionality, use the Functional Tests section, page 3-20.

### Switch/Jumper Setup Troubleshooting

Many problems can be traced to improper switch settings, jumper configurations, or backplane connections. Refer to Table 3-5 for possible problems caused by error made during installation and the corresponding corrective action(s).

Table 3-5 *Installation Troubleshooting Procedures*

Possible Problem	Corrective Action:
One or more switch S100 stations are in closed position.	Make certain switch stations are set to open position. (Closed switch station is indicated by blinking green LED on CSCI module.)
Switch S101 improperly set.	Make certain setting for all switch stations matches intended mode. (Switch settings for applicable modes are listed in "CSCI Module Installation" paragraph on page 3-17.)

### Audio Path Troubleshooting

Use Table 3-6 to perform audio path troubleshooting.

Table 3-6 *Audio Path Troubleshooting Procedures*

Problem	Corrective Action
No audio (test tone in, but no test tone out)	• Check all audio input and output connections at punchblock(s).
	• Make certain setting for all switch stations matches intended mode. (Switch settings for applicable modes are listed in "CSCI Module Installation" paragraph on page 3-17.)
	• Make certain 50-pin telco cable is properly mated to card cage and punchblock connectors.
	• On CSCI switch S100, make certain stations 1 and 2 are set to open position.
	• Perform tests described in "Audio Output Level Calibration," page 3-22 and "Maximum Output Level Verification," page 3-22.
Audio overdeviation (overmodulation) at transmitter	Perform procedure described in "Adjusting Audio Overdeviation," page 3-20.

**Troubleshooting****Adjusting Audio Overdeviation**

If overdeviation exists at the transmitter, perform adjustment as follows:

1. Verify the transmitter's in-cabinet repeat option is not active.
  - 1.1 Inject a +10 dBm, 1 kHz sinewave into the CSCI. The audio path should remain intact.
  - 1.2 Key the channel with a radio and talk over the test tone.
  - 1.3 Monitor the transmit frequency with a service monitor. The transmitter deviation should remain constant, near  $\pm 5$  kHz.
  - 1.4 If the deviation varies with the speech, the transmitter's in-cabinet repeat option is active. Disable the in-cabinet repeat option in the base station codeplug.
2. With the +0dBm, 1 kHz sinewave sent to the CSCI, monitor the transmitter (without voice) with a service monitor.
3. If the sinewave causes overdeviation, reduce deviation at the transmitter.

If overdeviation cannot be prevented as described in step 3, check power supply voltage and reduce as necessary. Refer to "CSCI Power Supply Connections and Adjustment," page 3-15, for adjustment procedure.

**CAUTION**

Do not reduce power supply voltage to less than 12V.  
Improper operation may result if power supply voltage is too low.

**Functional Tests****NOTE**

Perform the following procedures only if improper operation of the CSCI has been indicated.

**NOTE**

The following procedures are intended only as an aid to field personnel in performing calibrations, or identifying problems to be repaired by Motorola System Support Center (SSC). Field repair of the modules is not authorized. Any problems discovered during the course of the following procedures should be noted and reported to SSC when the module(s) are forwarded for repair.

**Test Setup General Requirements**

Follow these general guidelines when performing the functional tests.

- ☐ Set CSCI switches S100 and S101 to the settings specified in each procedure. Switch S100 is used to put the CSCI into four different test modes.
- ☐ When one or more stations on switches S100 or S101 are closed, the power LED should blink. If the LED does not blink, the on-board clock is probably inoperative. If this is noticed, stop testing and send the module to SSC for repair.
- ☐ Make sure the module is out of service and not connected to any other part of the system, except as specified in the functional test procedures listed in the following sections.
- ☐ Disconnect the audio inputs and outputs at the punchblock.
- ☐ Each procedure performs a check of the input and output for each major circuit of the card. If a fault indication is noted, perform the following:
  - ☐ Make sure the injected signals, connections, and terminations are correct.
  - ☐ Take the measurement again.
  - ☐ Send the module to SSC for repair, if the fault indication is verified. Include information about the problem and tests that have been performed with the respective results.

**Test Equipment Required**

- ☐ Transmission Test Set—HP3551A (or equivalent)
- ☐ Dual-channel oscilloscope with invert and add capability—Tektronix 464 Storage Scope or 465B Oscilloscope (or equivalent)
- ☐ Volt-Ohmmeter (VOM)—Fluke 8010A Digital Multimeter (or equivalent)
- ☐ Audio oscillator—HP204D (or equivalent)

**Troubleshooting****Audio Output Level Calibration**

Perform CSCI audio output calibration as follows:

1. Using audio oscillator, inject a -10 dBm, 1 kHz signal to the CSCI audio inputs (J1A-2 and J1A-3).
2. On CSCI switch S100, set station 2 to closed position (mute PL/DPL path).
3. Set transmission test set for 600 $\Omega$  termination load.
4. Connect transmission test set to the CSCI audio outputs (J1A-5 and J1A-4).
5. Adjust R112 on CSCI for a -10 dBm signal, as indicated by transmission test set. Send CSCI to SSC if reading cannot be obtained.
6. Connect channel 1 of the oscilloscope to J1A-5, Audio Out (+), and channel 2 of the oscilloscope to J1A-4, Audio Out (-).
7. Invert channel 2, and add channel 1 to channel 2. Oscilloscope should display approximately 700 mVp-p undistorted sine wave. Send CSCI to SSC if proper indication not observed.
8. Disconnect all test equipment.
9. Return switch S100-2 to the open position.
10. Proceed to "Maximum Output Level Verification."

**Maximum Output Level Verification**

Perform maximum output level verification as follows:

1. Using VOM, measure DC voltage across J1A-12 and J1A-1 on the CSCI interconnect board.
  - 1.1 If reading is 13.8 VDC, proceed to step 2.
  - 1.2 If reading is not 13.8 VDC, adjust R107 on power supply module for 13.8 VDC. If reading cannot be obtained, send power supply module to SSC for repair.
2. Using audio oscillator, inject a +10 dBm, 1 kHz signal to the CSCI audio inputs (J1A-2 and J1A-3).
3. On CSCI switch S100, set station 2 to closed position (mute PL/DPL path).
4. Set transmission test set for 600 $\Omega$  termination load.
5. Connect transmission test set to the CSCI audio outputs (J1A-5 and J1A-4).
6. Connect channel 1 of the oscilloscope to J1A-5, Audio Out (+), and channel 2 of the oscilloscope to J1A-4, Audio Out (-).
7. Invert channel 2, and add channel 1 to channel 2. Observe oscilloscope.
  - 7.1 If oscilloscope displays 740 mVp-p trapezoidal waveform, proceed to step 8.

**CAUTION**

In next step, do not reduce power supply voltage to less than 12 VDC. Improper operation may result if power supply voltage is too low.

- 7.2 If oscilloscope displays amplitude greater than 740 mVp-p trapezoidal waveform, adjust R107 on power supply power supply for specified audio output amplitude. Send CSCI to SSC if you do not observe proper indication.
- 7.3 If waveform does not appear, or cannot be brought into specification using the previous steps, send CSCI module to SSC for repair. Recalibrate power supply as shown in steps 7 through 7.2 of this section.
8. Disconnect all test equipment.
9. Return switch S100-2 to the open position.
10. Proceed to “Pre-Emphasis Calibration.”

**Pre-Emphasis Calibration**

Perform pre-emphasis calibration as follows:

**NOTE**

In the following procedure, if any reading cannot be adjusted into specified range, send CSCI module to SSC for repair.

1. Using audio oscillator, inject a -20 dBm, 1 kHz signal to the CSCI audio inputs (J1A-2 and J1A-3).
2. On CSCI switch S100, set station 2 to closed position (mute PL/DPL path).
3. Set transmission test set for 600 $\Omega$  termination load.
4. Connect transmission test set to the CSCI audio outputs (J1A-5 and J1A-4).
5. On CSCI, adjust R112 for a -20 dBm output level, as indicated by transmission test set.
6. Adjust audio oscillator for the frequencies listed in the following table. Using transmission test set, verify approximate output level for each frequency.

**Troubleshooting**

Frequency	Level
300 Hz	-30.46 dB (+1/-3 dB)
600 Hz	-24.46 dB
1.2 kHz	-18.46 dB
3 kHz	-12.8 dB
6 kHz	-16.5 dB
12 kHz	-22.5 dB
24 kHz	<-35 dB

7. Disconnect all test equipment.
8. Return switch S100-2 to the open position.
9. Proceed to 2:1 Compression Calibration.

**2:1 Compression Calibration**

Perform 2:1 compression calibration as follows:

**NOTE**

In the following procedure, if any reading cannot be adjusted into specified range, send CSCI module to SSC for repair.

1. Using audio oscillator, inject a -8.1 dBm, 1 kHz signal to the CSCI audio inputs (J1A-2 and J1A-3).
2. On CSCI switch S101, set station 2 to open position (enable compression).
3. On CSCI switch S100, set station 2 to closed position (mute PL/DPL path).
4. Set transmission test set for 600 $\Omega$  termination load.
5. Connect transmission test set to the CSCI audio outputs (J1A-5 and J1A-4).
6. On CSCI, adjust R112 for a -10 dBm output level, as indicated by transmission test set.
7. On audio oscillator, decrease signal level by 2.0 dBm. The output level indicated on transmission test set should correspondingly decrease by 1.0 dBm (+0.1 dBm).
8. Decrease the oscillator level by an additional 2.0 dBm. The output level indicated on transmission test set should now read -12 dBm (+0.2 dBm).

9. On audio oscillator, set signal level to -11.94 dBm. The output level indicated on transmission test set should now read -11.94 dB (+0.1 dBm).
10. On CSCI switch S101, set station 2 to closed position (disable compression).
11. Set audio oscillator for a -10 dBm signal level.
12. On CSCI, adjust R112 for a -10 dBm output level, as indicated by transmission test set.
13. Remove all test equipment.
14. Return switch S100-2 to the open position.
15. Proceed to Coded Indicate Verification.

### Coded Indicate Verification

Perform coded indicate verification as follows:

<b>NOTE</b>
-------------

If any of the normal indications specified in the following procedure are not observed, send CSCI module to SSC for repair.

1. Remove power from the CSCI card cage.
2. Using a test clip, ground J1A-6.
3. Using test clips, temporarily connect a 10 k $\Omega$  resistor between J1A-7 and J1A-12.
4. Apply power to the CSCI card cage.
5. Using a VOM, measure the voltage across J1A-7 and J1A-1. Verify voltage is greater than 12V.
6. Using a test clip, ground J1A-15 (Coded Indicate In).
7. Observe VOM. The voltage should now be less than 1 V.
8. Remove power from the CSCI card cage.
9. Remove all test clips.
10. Remove all test equipment.
11. Apply power to the CSCI card cage.
12. Proceed to PTT In/Out Verification section.

**Troubleshooting****PTT In/Out Verification**

Perform PTT in/out verification as follows:

**NOTE**

If any of the normal indications specified in the following procedure are not observed, send CSCI module to SSC for repair.

1. Remove power from the CSCI card cage.
2. Using a test clip, ground J1A-11.
3. Using test clips, temporarily connect a 10 k $\Omega$  resistor between J1A-12 and J1B-1.
4. Apply power to the CSCI card cage.
5. Using a VOM, measure the voltage across J1A-12 and J1B-1. Verify voltage is less than 1V.
6. Using a test clip, ground J1A-10 and verify the red LED illuminates.
7. Using a VOM, measure the voltage across J1A-12 and J1B-1. Verify voltage is greater than 12V.
8. Remove all test equipment.
9. Proceed to "Audio Mute Verification".

**Audio Mute Verification**

Perform audio mute verification as follows:

**NOTE**

If any of the normal indications specified in the following procedure are not observed, send CSCI module to SSC for repair.

1. Using transmission test set, inject a -10 dBm, 1 kHz signal to the Audio inputs at J1A-3 and J1A-2.
2. On CSCI switch S100, set station 1 to closed position (Mute Audio Path).
3. On CSCI switch S100, set station 2 to open position (enable PL/DPL path). Using an oscilloscope, monitor the audio output at J1A-5 and J1A-4. Only the PL/DPL signal should appear at the output.



4. On CSCI switch S100, set station 1 to open position (enable PL/DPL path). Both the 1 kHz and PL/DPL signals should now appear at the output.
5. Remove all test equipment.
6. Proceed to PL/DPL Output Level Calibration section.

### PL/DPL Output Level Calibration

Perform data output level calibration as follows:

<b>NOTE</b>
-------------

In the following procedure, if any reading cannot be adjusted into specified range, send CSCI module to SSC for repair.

1. On CSCI switch S100, set station 1 to closed position and station 2 to open position.
2. Connect Transmission Test Set across CSCI audio output at J1A-5 and J1A-4.
3. On CSCI, adjust R141 for a 61.5 mVp-p (-22 dBm) level, as observed on Transmission Test Set.
4. Check deviation (modulation) for the output levels listed in the following table.

Deviation	Output (dBm)	Output (mV)
1 kHz	-19.54	82.0
750 Hz	-22.00	61.5
500 Hz	-25.56	41.0

5. Return all S100 switches to open position.
6. Remove all test equipment.

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**Troubleshooting**

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## CSCI (QTN6066B) Parts List

## CSCI (QTN6066B) Parts List

Reference	Part Number	Description
<b>capacitor, fixed:</b>		
C100 thru 102	0811051A19	1uF, +5%/-0.5%; 63V
C103	2113740B57	220 pF, ±30%; 50V
C104	0811051A19	1uF, +5%/-0.5%; 63V
C105	2311049A10	2.2uF, ±10%; 35V
C106,107	0811051A19	1uF, +5%/-0.5%; 63V
C108	2113741B37	4700pF, ±5%; 50V
C109	2113740B56	200pF, ±5%; 50V
C110	2113740B57	220pF, ±30%; 50V
C111	2113740B73	1000pF, ±5%; 50V
C112	2113740B80	2200pF, ±5%; 50V
C113,114	2313748G19	47uF, ±20%; 35V
C115	2113741B45	0.01uF, ±5%; 50V
C116	2311049A15	4.7uF, ± 10%; 35V
C117	2113740B63	390pF, ±5%; 50V
C118	2113740B67	560pF, ±5%; 50V
C119	2113740B49	100pF, ±5%; 50V
C120 thru 123	2113740B73	1000pF, ±5%; 50V
C124	2113741B69	0.1uF, ±5%; 50V
C125,126	2311049A20	15uF, ±10%; 20V
C127	2113741B45	0.01uF, ±5%; 50V
C128	2113740B57	220pF, ±30%; 50V
C129	2311049A20	15uF, ± 10%; 20V
C130,131	2313748G22	100uF, ±20%; 25V
C132	2311049A20	15uF, ±10%; 20V
C133	2113740B73	1000 pF, ±5%; 50V
C134	2311049A21	22uF, ±10%; 20V
C135	2113740B73	1000pF, ±5%; 50V
C136	2311049A08	1uF, ±10%; 35 V
C137,138	2113740B73	1000pF, ±5%; 50V
C139	2113740B68	620pF, ±5%; 50V
C140	2113741B45	0.01uF, ±5%; 50V
C141	2113741 B37	4700pF, ±5%; 50V
C142	2113740B32	20pF, ±5%; 50V
C143	2113740B49	100pF, ±5%; 50V
C144	2113740B59	270pF, ±5%; 50V
C145	2113741B45	0.01uF, ±5%; 50V
C146	0811051A19	1uF, ±5%/-0.5%; 63V
C147,148	2113741B57	0.033uF, ±5%; 50V
C149	2113740B73	1000pF, ±5%; 50V
C150	2113741B33	3300pF, ±5%; 50V
C151	2311049A20	15uF, ± 10%; 20V
C152	2113741 B41	6800pF, ±5%; 50V
C153	2113741B39	5600pF, ±5%; 50V
C154	2113741B33	3300pF, ±5%; 50V
C155	0811051A16	0.33uF, ±5%; 63V
C156	2113741B23	1200pF, ±5%; 50V
C157	0811051A14	0.15uF, ±5%; 63V
C158	2311049A20	15uF, ±10%; 20V
C159	2113741B35	3900 pF, ±5%; 50V
C160 thru 169	2113741 B69	0.1uF, ± 5%; 50 V
C170	2113741B45	0.01uF, ±5%; 50V
C171*	2311049A17	6.8uF, ±10%; 35V
C172*	2311049A20	15uF, ±10%; 20V
<b>diode: (see note)</b>		
CR100	4813833C04	dual 70V
CR101	4813833C07	dual 100W
CR102	4813833C03	dual 70V
CR103	4813833C07	dual 100W
CR104	4813833C10	0.1A, 70V
CR105 thru 110	4813833C07	dual 100W
CR111 thru 114	4813833C10	0.1A, 70V

Reference	Part Number	Description
CR115	4813833C03	dual 70V
CR116,117	4813833C10	0.1A, 70 V
CR118	4813833C04	dual 70V
CR119	4813833C10	0.1A, 70V
CR120	4813833C07	dual 100W
CR121	4813833C04	dual 70V
CR122	4813833C03	dual 70V
CR123	4813833C10	0.1A, 70V
CR124	4813833C07	dual 100W
CR125 thru 129	4813833C10	0.1A, 70V
CR130* thru 132*	4813833C10	0.1A, 70V
<b>light emitting diode: (see note)</b>		
DS1	4888245C22	GRN
DS2	4888245C24	RED
DS3	4888245C23	YEL
<b>fuse:</b>		
F1	6582408R05	1/2A, 125V
<b>connector:</b>		
J1A	0983445L11	receptacle: 12-contact
J1 B	0983445L11	receptacle: 12-contact
J2,3	0984271L03	receptacle: phone jack
<b>transistor (see note)</b>		
Q100 thru 103	4813824A10	NPN
Q104	4800869652	type N-C JFET
Q105,106	4813824A10	NPN
Q107	4813824A17	PNP
<b>resistor, fixed:</b>		
R100	0611077E73	604 ohms, ±1%; 1/8W
R101,102	0611077G57	47.5K, ±1%; 1/8W
R103,104	0611077F24	2K, ±1%; 1/8W
R105,106	0611077B15	47K, ±5%; 1/8 W
R107 thru 110	0611077F24	2K, ±1%; 1/8W
R111	0611072A49	1K, ±5%, 1/4W
R112	1883452F17	variable: 50K, ±10%; 1/2W
R113	0611077F71	6.19K, ±1%; 1/8W
R114	0611077A92	5.6K, ± 5%; 1/8W
R115	0611077G68	61.9K, ±1%; 1/8W
R116	0611077B11	33K, ±5%; 1/8W
R117	0611077B06	20K, ±5%; 1/8W
R118	0611077A01	0 ohm, ±5%; 0W
R119	0611077B16	51 K, ±5%; 1 /8W
R120	0611077B13	39K, ±5%; 1/8W
R121	0611077B15	47K, ±5%; 1/8W
R122	0611077B13	39K, ±5%; 1/8W
R123,124	0611077B15	47K, ±5%; 1/8 W
R125	0611077F28	2.21K, ±1%; 1/8W
R126	0611077A82	2.2K, ±5%; 1/8W
R127	0611077G68	61.9K, ±1%; 1/8W
R128	0611077B31	220K, ±5%; 1/8 W
R129	0611077B15	47K, ±5%; 1/8W
R130	0611077A82	2.2K, ±5%; 1/8 W
R131	0611077B15	47K, ±5%; 1/8W
R132	0611077B31	220K, ±5%; 1/8 W
R133	0611077B13	39K, ±5%; 1/8W
R134	0611077B15	47K, ±5%; 1/8W

**CSCI (QTN6066B) Parts List**

Reference	Part Number	Description	Reference	Part Number	Description
R135	0611077B01	12K, $\pm 5\%$ ; 1/8W	R231	0680149M02	100K, $\pm 10\%$ ; 240 MW
R136	0611077B03	15K, $\pm 5\%$ ; 1/8W	R232	0611077B07	22K, $\pm 5\%$ ; 1/8W
R137	0611077A94	6.8K, $\pm 5\%$ ; 1/8W	R233	0611077A89	4.3K, $\pm 5\%$ ; 1/8W
R138	0611077B15	47K, $\pm 5\%$ ; 1/8W	R234	0611077B23	100K, $\pm 5\%$ ; 1/8W
R139	0611077G25	22.1K, $\pm 1\%$ ; 1/8W	R235	0611077A96	8.2K, $\pm 5\%$ ; 1/8W
R140	0611077B15	47K, $\pm 5\%$ ; 1/8 W	R236	0611077F77	7.15K, $\pm 1\%$ ; 1/8W
R141	1883452F19	variable: 100K $\pm 10\%$ ; 1/2W	R237	0611077B23	100K, $\pm 5\%$ ; 1/8 W
R142	0611077B05	18K, $\pm 5\%$ ; 1/8W	R238	0611077H04	143K, $\pm 1\%$ ; 1/8W
R143	0611077G76	75K, $\pm 1\%$ ; 1/8 W	R239	0611077F28	2.21K, $\pm 1\%$ ; 1/8W
R144	0611077B15	47K, $\pm 5\%$ ; 1/8W	R240	0611077A93	6.2K, $\pm 5\%$ ; 1/8W
R145 thru 147	0611077H04	143K, $\pm 1\%$ ; 1/8W	R241	0611077A89	4.3K, $\pm 5\%$ ; 1/8W
R148	0611077G88	100K, $\pm 1\%$ ; 1/8W	R242,243	0611077G34	27.4K, $\pm 1\%$ ; 1/8W
R149	0611077H04	143K, $\pm 1\%$ ; 1/8W	R244 thru 246	0611077A98	10K, $\pm 5\%$ ; 1/8W
R150	0611077G76	75K, $\pm 1\%$ ; 1/8W	R247*,248*	0611077A66	470, $\pm 5\%$ ; 1/8W
R151	0611077B10	30K, $\pm 5\%$ ; 1/8W	R249*	0611077A01	0 ohm, $\pm 5\%$
R152	0611077B15	47K, $\pm 5\%$ ; 1/8W			<b>switch:</b>
R153	0611077A92	5.6K, $\pm 5\%$ ; 1/8W			
R154	0611077F60	4.75K, $\pm 1\%$ ; 1/8W	S100	4083849F09	rocker: 4-position
R155	0611077A90	4.7K, $\pm 5\%$ ; 1/8W	S101	4083849F04	rocker: 4-position
R156	0611072A49	1 K, $\pm 5\%$ , 1/4W	S102,103	4083849F10	rocker: 10-position
R157	0611077F60	4.75K, $\pm 1\%$ ; 1/8W			<b>integrated circuit:</b>
R158,159	0611077G57	47.5K, $\pm 1\%$ ; 1/8W			<b>(see note)</b>
R160	0611077B15	47K, $\pm 5\%$ ; 1/8W			
R161	0611077G57	47.5K, $\pm 1\%$ ; 1/8W			
R162	0611072A49	1K, $\pm 5\%$ , 1/4W			
R163	0611077B15	47K, $\pm 5\%$ ; 1/8W	U100	5182276R48	Low-Noise, JFET-Input Operational Amplifier
R164	0611072A49	1K, $\pm 5\%$ , 1/4W	U101	5184887K73	Quad Bilateral Switch
R165	0611077F91	10K, $\pm 1\%$ ; 1/8W	U102	5182276R48	Low-Noise, JFET-Input Operational Amplifier
R166	0611072A49	1K, $\pm 5\%$ , 1/4W			
R167	0611077F91	10K, $\pm 1\%$ ; 1/8W	U103	5184887K60	Triple 2-Channel Analog Mux/Demux
R168,169	0611077E15	150 ohms, $\pm 1\%$ ; 1/8W			
R170 thru 173	0611077A42	47 ohms, $\pm 5\%$ ; 1/8W			
R174,175	0611077E15	150 ohms, $\pm 1\%$ ; 1/8W	U104,105	5182276R48	Low-Noise, JFET-Input Operational Amplifier
R176,177	0611077F91	10K, $\pm 1\%$ ; 1/8W			
R178	0611072A37	330 1/4W	U106	5184621K86	Dual Gain Control
R179	0611072A03	12 ohms, $\pm 5\%$ ; 1/4W	U107,108	5184887K01	Inverting Hex Buffer
R180	0611077B15	47K, $\pm 5\%$ ; 1/8W	U109	5184320A35	Timer
R181	0611072A32	200 ohms, $\pm 5\%$ ; 1/4W	U 110,111	5184320A92	Optoelectronic Isolator
R182,183	0611077B15	47K, $\pm 5\%$ ; 1/8W	U 112	5183977M 17	PLED Encoder
R184	0611077A90	4.7K, $\pm 5\%$ ; 1/8W	U113	5180067C03	Dual Operational Amplifier
R185	0611077A98	10K, $\pm 5\%$ ; 1/8W	U 114	5184267A82	CMOS Digital Detector
R186	0611077B23	100K, $\pm 5\%$ ; 1/8W			<b>voltage regulator:</b>
R187	0611077A98	10K, $\pm 5\%$ ; 1/8W			<b>(see note)</b>
R188	0611072A42	510 ohms, $\pm 5\%$ ; 1/4W			
R189,190	0611077B23	100K, $\pm 5\%$ ; 1/8W	VR100	4882479V01	Zener 2.7V
R191	0611077B11	33K, $\pm 5\%$ ; 1/8W	VR101	4882479V05	Zener 6.2V
R192 thru 195	0611077B23	100K, $\pm 5\%$ ; 1/8W	VR102	4882479V09	Zener 9.1V
R196	0611077B09	27K, $\pm 5\%$ ; 1/8W			<b>crystal: (see note)</b>
R197 thru 202	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R203	0611077B39	470K, $\pm 5\%$ ; 1/8W			
R204	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R205	0611077B29	180K, $\pm 5\%$ ; 1/8W	Y100	4880055C01	Tuning Fork
R206	0611077A98	10K, $\pm 5\%$ ; 1/8W	Y101	4882003K01	Resonator
R207 thru 209	0611077B15	47K, $\pm 5\%$ ; 1/8W			<b>non-referenced items:</b>
R210	0611077B03	15K, $\pm 5\%$ ; 1/8W			
R211	0611077A89	4.3K, $\pm 5\%$ ; 1/8W			
R212 thru 214	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R215	0611077B11	33K, $\pm 5\%$ ; 1/8W			
R216	0611077B39	470K, $\pm 5\%$ ; 1/8W			
R217,218	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R220,221	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R222	0611077B10	30K, $\pm 5\%$ ; 1/8W			
R223	0611077B07	22K, $\pm 5\%$ ; 1/8 W			
R224	0611077F41	3.01K, $\pm 1\%$ ; 1/8W			
R225	0680149M01	470 ohms, $\pm 10\%$ ; 240 MW			
R226	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R227	0680149M02	100K, $\pm 10\%$ ; 240 MW			
R228	0611077B23	100K, $\pm 5\%$ ; 1/8W			
R229	0611077B15	47K, $\pm 5\%$ ; 1/8W			
R230	0611077B23	100K, $\pm 5\%$ ; 1/8W			

**NOTE 1:** FOR OPTIMUM PERFORMANCE, DIODES, TRANSISTORS, INTEGRATED CIRCUITS, AND CRYSTALS MUST BE ORDERED BY MOTOROLA PART NUMBER.

**NOTE 2:** COMPONENTS SHOWN WITH AN ASTERISK (\*) USED ON QTN6066B VERSION ONLY.



# ***Simulcast Distribution Amplifier***

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This chapter provides a description of the Simulcast Distribution Amplifier (SDA), model number QLN3174A. The following topics are included in this chapter:

Topic	Page	Description
Description	4-2	Describes the role of the SDA in a simulcast system.
Theory of Operation	4-4	Describes the SDA circuitry.
Troubleshooting	4-7	Provides SDA troubleshooting information and corrective action.
Switch Settings Reference Guide	4-8	Lists SDA switch settings for typical configuration setups.
SDA Edge Connector Pinout	4-9	Lists the SDA edge connector signals and cross-references connector pins to telco punchblock connector pins.
SDA Parts List/Schematic Diagram	4-11	Provides the SDA schematic diagram and parts list.

**Description**

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## Description

The distribution amplifier board (QRN4878A) within the SDA splits 600 $\Omega$  transmit audio from a Universal Simulcast Controller Interface (USCI) or Conventional Simulcast Controller Interface (CSCI) into ten equal 600 $\Omega$  line outputs with no net loss. Each 600 $\Omega$  line output is intended to drive a 600 $\Omega$  line input of a DSM/DSM-II destined for a different simulcast transmitter. However, any 600 $\Omega$  balanced line may be driven, including the input of another SDA.

The distribution amplifier accepts a nominal line input test tone level of -10 dBm and delivers -10 dBm at each of the ten output lines. The amplifier is virtually noiseless and distortionless as required by the simulcast system application, and provides extended low frequency response for low frequency tone transmissions such as Private Line (PL), or low speed digital transmissions such as Digital Private Line (DPL) or Trunked System Low Speed Data. Group Delay Distortion is negligible as required for such high-speed digital transmissions as the Trunked System High Speed Data signals.

If more than ten simulcast transmitters are on the same channel, one of the ten 600 $\Omega$  line outputs should be used to drive the 600 $\Omega$  input line of another distribution amplifier. This results in a capacity of 19 lines. If more than 19 line outputs are required, the process may be repeated again, resulting in a total capacity of 28 600 $\Omega$  line outputs for delivery to as many as 28 DSM Channel Unit 600 $\Omega$  line inputs (see Figure 4-1).

In general, not all of the 600 $\Omega$  line outputs will be utilized in a given system configuration. In this case, each unused line should be terminated in a 600 $\Omega$  resistive load which is provided on the distribution amplifier module.

The distribution amplifier can also split two signaling lines, each ten ways, for distribution to the remote sites. Typically, these will be transmitter PTT lines.

Front accessible DIP switches on the distribution amplifier are provided for manually generating PTT (or other signal) and also for disabling any individual PTT (or other signal) line. This feature is of value during alignment and testing of the simulcast system.

If more than ten simulcast transmitters are on the same channel, the PTT (MLead) splitters must be cascaded by wiring the common output of the first module to the input of the next, and so on (see Figure 4-1).

In a trunked system, the PTT line is generated in the central controller. In a nontrunked system, the PTT line is generated by the voting equipment.

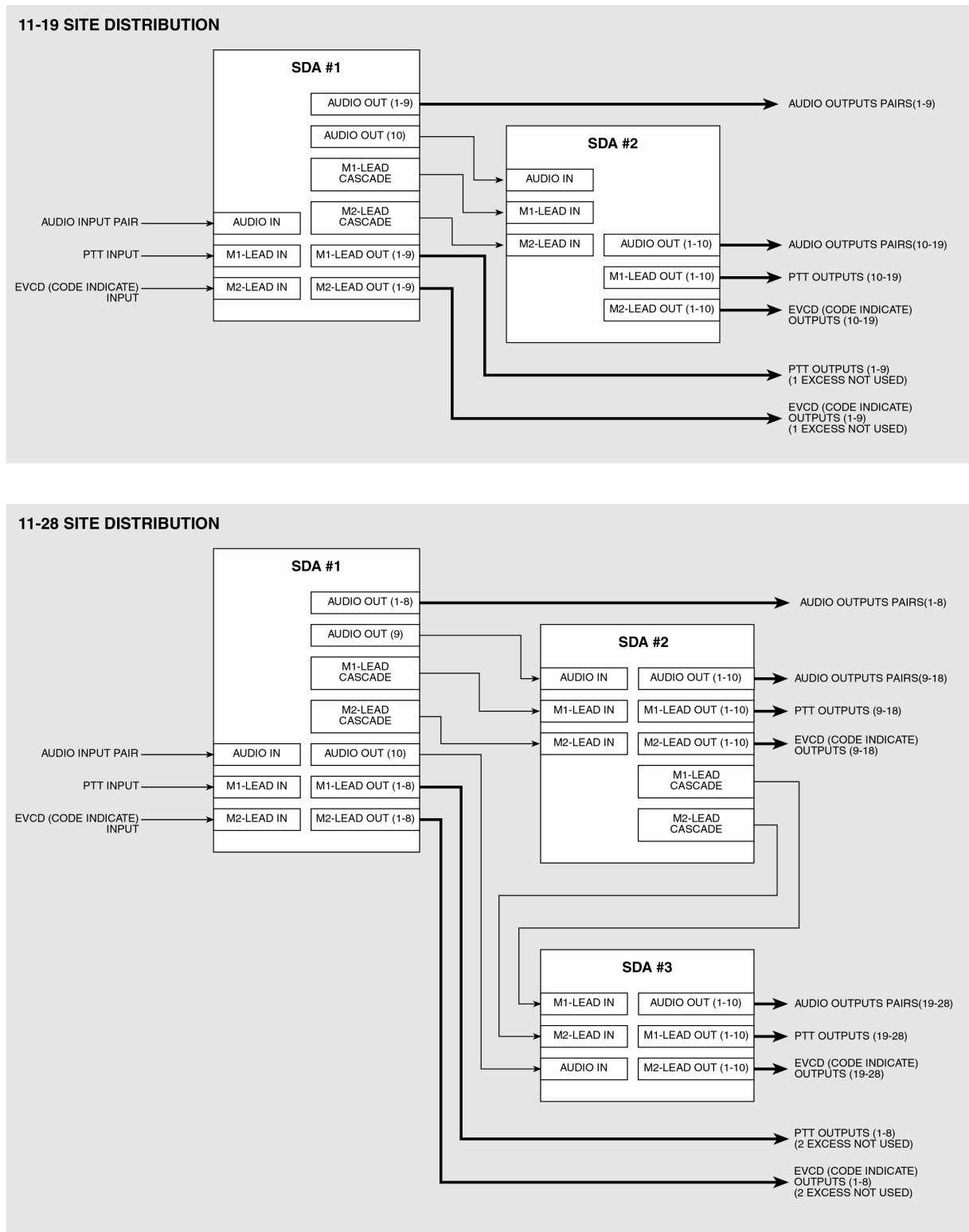
SIMU039  
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Figure 4-1 SDA Interconnections (more than 10 sites)

## Theory of Operation

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## Theory of Operation

The following sections describe the SDA circuits. Refer to the SDA schematic diagram shown in Figure 4-2.

### Regulator

Regulated power is provided to op amp U303 and Push-to-Talk (PTT) and Encrypted Voice Code Detect (EVCD) (also called Code Indicate) switching circuits via two negative voltage regulators (U301 and U302). U301 utilizes feedback resistors R301 and R302 to regulate the -42 to -56 V(-48 VDC nominal) battery potential to -18 VDC. The -18 VDC supply is applied to three different circuits.

- ❑ -18 VDC is applied to another regulator circuit comprised of U302, R303, and R304. This second regulator converts the -18 VDC input to a -9 VDC output. The -9 volt output of the second regulator is also applied to pins 3 and 12 of op amplifier U303, pins 6 and 7 of switch S301, and to several nodes on signaling circuits PTT and EVCD (Code Indicate) in order to power those circuits.
- ❑ -18 VDC is applied to pin 11 of op amp U303.
- ❑ -18 VDC and -9 VDC from the output of the second regulator is applied to the Regulator Operate LED.

### Regulator Operate LED

The regulator operate circuitry is designed to detect a failure in either or both of the regulator circuits. If both regulators are operating near their nominal voltages, the green LED (DS401) is illuminated. The LED will be extinguished if the -18 VDC supply rises beyond -14 VDC or if the -9 VDC supply rises beyond -6.7 VDC.

### Audio Amplifier/Splitter

The audio amplifier/splitter (labeled DISTRIBUTION AMPLIFIER on the schematic) is a balanced, non-inverting amplifier that uses two op amp sections of U303. The input audio signal enters the board through pins 41 and 42 and is loaded by R331 (604 $\Omega$ ) to match the output impedance of the driving device. The audio signal then passes through positions 1 and 4 on front accessible switch S301, and through coupling capacitors C306, C307, C309 and C310 before reaching the noninverting inputs of the op amps. Positions 1 and 4 of S301 allow a user to interrupt the audio to the amplifier. The coupling capacitors are loaded with 499k $\Omega$  resistors R305 and R318 to provide an AC frequency response high-pass corner at about 0.016 Hz. Feedback resistors R306, R307, R319 and R320 provide a voltage gain of two between the output and non-inverting input of the op amps.



Since the output of the op amps is a low impedance signal source, ten  $301\Omega$  resistors are between the op amp output and each of the ten output pins in each of the two halves of the balanced amplifier. These resistors ensure that the overall amplifier has unity gain and a  $600\Omega$  output impedance to match the  $600\Omega$  input impedance of the DSM/DSM-II card.

If all ten outputs of the splitter are not used, the unused outputs should be terminated with  $600\Omega$ s. A load of  $604\Omega$ s is supplied for each output and can be switched in by closing the appropriate switch position on S302.

One distribution amplifier can drive up to ten DSM/DSM-II cards. There is one DSM-II card per simulcast site, per audio channel. If an audio channel has more than ten simulcast sites, one or more outputs of the driving distribution amplifier can be used to drive other amplifiers.

## Signaling Circuits

The distribution amplifier board has two signaling/relay circuits (M-leads) labeled PTT and EVCD (Code Indicate). The two circuits are identical to each other in operation and composition and are designed to be keyed in either of the following two ways:

1. Shorting the two input leads together either by a hard short-circuit closure as in Spectra-TAC™, or by a saturated opto-isolator output as in the Simulcast Digital Microwave Interface.
2. A high (+13.8 V) potential falling to a low (approximately equal to 0 V) potential as in DIGITAC.

If the driving device uses a short circuit or saturated opto-isolator closure, positions 1 and 2 of S102 and S202 should be closed. If the driving device uses a high-potential to low-potential key, then positions 1 and 2 of S102 and S202 should be open. Positions 3 and 4 of S102 and S202 are not connected.

The output of the signaling circuit is intended to drive the input of the M-leads on the DSM board only.

The operation of the signaling circuits is as follows. R103 (R203) and R108 (R208) form a voltage divider that holds the potential of Q101 (Q201) emitter at about -1 V. When the circuit is unkeyed, Q101 (Q201), Q102 (Q202), and Q103 (Q203) are all cut off. When the circuit is keyed, all the transistors are saturated. The approximate potential at each of the transistor terminals for each state is listed in Table 4-1.

## Theory of Operation

Table 4-1 *Transistor/Terminal Voltage Potential*

Transistor/Terminal Voltage	Keyed	Unkeyed
Q101 (Q201)		
Vb	-1.7V	-1.0V (Spectra-TAC, central controller) +2.8V (DIGITAC)
Ve	-1.1V	-1.0V
Vc	-1.1V	-9.2V
Q102 (Q202)		
Vb	-8.4V	-9.2V
Ve	-9.2V	-9.2V
Vc	-9.1V	-0.1V
Q103 (Q203)		
Vb	-0.72V	-0.07V
Ve	0.0V	0.0V
Vc	-0.04V	-4.5V

The three front accessible switches S101, S201, and S301 (positions 2 and 3) enable the user to have manual control of the signaling. Both S101 and S201 are ten position switches, each position of which can prevent or allow its corresponding site to be keyed depending on whether the switch position is open or closed, respectively. S101 controls PTT (or M-lead 1), and S201 controls EVCD (Code Indicate, or M-lead 2). Note that even if a given switch position is closed, its site will be keyed only if the input to the signaling circuit is keyed. The user can manually force either signaling circuit to be keyed by closing positions 2 or 3 of S301. S301 position 2 keys the PTT or M-lead 1 and S301 position 3 keys the EVCD (Code Indicate) or M-lead 2.

Each of the two signaling circuits can key the appropriate M-lead input in as many as ten sites. If a system has more than ten sites per audio channel, pin 47 of the PTT circuit and pin 48 of the EVCD (Code Indicate) on the primary distribution amplifier can be connected to pins 43 and 44, respectively, of the secondary distribution amplifier board. This cascading of boards is possible if S102 and S202 positions 1 and 2 are set to “closed” on the secondary board, regardless of the setting of those switches on the primary board. If there are more than 19 simulcast sites in a system, the signaling inputs on the tertiary card must be driven by pins 47 and 48 of the secondary card, not the primary card. Note that this cascading scheme for the signaling circuits is different from that for the audio circuit.

## Troubleshooting

If you experience problems with the SDA, refer to Table 4-2.

Table 4-2 **SDA Troubleshooting**

Problem	Procedure
No audio, or low audio (3.54 dBm or under expected level) at the DSM/DSM-II input.	<ul style="list-style-type: none"> <li>• Make sure the audio signal is available from the USCI/CSCI or the voting equipment to the distribution amplifier audio input (pins 41 and 42).</li> <li>• If audio is present at the input, make sure that both positions 1 and 4 of switch S301 are closed so that the input signal can reach the amplifier.</li> <li>• If an audio signal is present at the DSM/DSM-II input, but is low in level (most likely by 3.54 dB), make sure that the position on switch S302 corresponding to the output in question is open. (If the switch is closed, a 604Ω load is connected in parallel with the 604Ω load of the DSM/DSM-II audio input. The additional 604Ω resistor causes excessive loading of the distribution amplifier output.)</li> </ul>
Unable to key either of the signaling circuits of the DSM/DSM-II board with the PTT or EVCD (Code Indicate) circuits.	<ul style="list-style-type: none"> <li>• Verify that the keying signal is available from the Spectra-TAC, DIGITAC, or the USCI/CSCI to the input pins of the distribution amplifier signaling circuit(s).</li> <li>• If the signal from the driver is present, verify that positions 1 and 2 of switches S102 and S202 are set properly.</li> <li>• Set positions 1 and 2 to closed if the driving device is a Spectra-TAC or USCI/CSCI.</li> <li>• If the driving device is a DIGITAC, set positions 1 and 2 to open.</li> <li>• If the distribution amplifier board signaling circuitry still will not transmit the keying signal to the DSM/DSM-II board at a given site, verify that the appropriate position on switch S101 for M-lead 1 (or PTT) or S201 for M-lead 2 (or EVCD/Code Indicate) is closed. (If the switch position in question is open, no signal can be transmitted to the site.)</li> </ul>
M-leads on the DSM/DSM-II boards remain keyed even when no keying signal is present from the voting equipment (Spectra-TAC, DIGITAC, or USCI/CSCI).	<ul style="list-style-type: none"> <li>• Make sure that positions 2 and 3 of switch S301 are in the open position. (If either or both of positions 2 and 3 of switch S301 are closed, the corresponding signaling circuit (PTT or EVCD/Code Indicate, or both) will be keyed on regardless of the signal from the voting equipment.)</li> <li>• If the DSM/DSM-II M-leads are still on when no triggering signal is being delivered from the voting equipment, verify that positions 1 and 2 of switches S102 and S202 are in the proper position for the type of voting equipment being used. (Positions 1 and 2 of switches S102 and S202 should be in the closed position if the driving device is a Spectra-TAC or USCI/CSCI and in the open position if the driving device is a DIGITAC.)</li> </ul>

## Switch Settings Reference Guide

## Switch Settings Reference Guide

Table 4-3 lists the proper SDA switch settings for typical configuration setups.

### NOTE

Refer to Figure 4-1 if the system uses more than 10 simulcast sites.

Table 4-3 **SDA Switch Settings (Typical Configurations)**

SDA Switch	Configuration	Switch Setting
S101, S201	Any position driving a DSM/DSM-II M-Lead input	Set to CLOSED
	Any position <b>not</b> driving a DSM/DSM-II M-Lead input – set to OPEN	Set to OPEN
S102, S202	If either one or both of the following conditions apply: <ul style="list-style-type: none"> <li>• DIGITAC is the triggering equipment</li> <li>• Key-up is triggered by a positive potential changing to GND on pin 43 for PTT (M1-Lead) and/or pin 44 for EVCD/Code Indicate (M2-Lead)</li> </ul>	Set positions 1 and 2 to OPEN  (Positions 3 and 4: don't care)
	If any of the following three conditions apply: <ul style="list-style-type: none"> <li>• Spectra-TAC, USCI, or CSCI is the triggering equipment</li> <li>• Key-up is triggered by a dead short-circuit or a saturated optoisolator across pins 43 and 45 for PTT (M1-Lead) and/or pins 44 and 46 for EVCD/Code Indicate (M2-Lead)</li> <li>• Key-up is triggered by another distribution amplifier (i.e., pin 47 of primary amplifier to pin 43 of secondary amplifier (PPT or M1-Lead) and pin 48 of primary amplifier to pin 44 of secondary amplifier (EVCD/Code Indicate or M2-Lead))</li> </ul>	Set positions 1 and 2 to CLOSED  (Positions 3 and 4: don't care)
S301	Any configuration	Set positions 1 and 4 to CLOSED  Set positions 2 and 3 to OPEN
S302	Any position driving a DSM/DSM-II or another distribution amplifier	Set to OPEN
	Any position <b>not</b> driving a DSM/DSM-II or another distribution amplifier	Set to CLOSED

## SDA Edge Connector Pinout

Table 4-4 lists the SDA edge connector signals. It also cross-references the edge connector pins to telco punchblock connector pins.

Table 4-4 **SDA Edge Connector Pinouts**

SDA Edge Connector Pin No.	Punchblock Pin No.	Signal Description
1	1	Audio out (ring); site 1
2	26	Audio out (tip); site 1
3	2	Audio out (ring); site 2
4	27	Audio out (tip); site 2
5	3	Audio out (ring); site 3
6	28	Audio out (tip); site 3
7	4	Audio out (ring); site 4
8	29	Audio out (tip); site 4
9	5	Audio out (ring); site 5
10	30	Audio out (tip); site 5
11	6	Audio out (ring); site 6
12	31	Audio out (tip); site 6
13	7	Audio out (ring); site 7
14	32	Audio out (tip); site 7
15	8	Audio out (ring); site 8
16	33	Audio out (tip); site 8
17	9	Audio out (ring); site 9
18	34	Audio out (tip); site 9
19	10	Audio out (ring); site 10
20	35	Audio out (tip); site 10
21	11	M1-Lead (PTT) output; site 1
22	36	M2-Lead (EVCD/Code Indicate) output; site 1
23	12	M1-Lead (PTT) output; site 2
24	37	M2-Lead (EVCD/Code Indicate) output; site 2
25	13	M1-Lead (PTT) output; site 3
26	38	M2-Lead (EVCD/Code Indicate) output; site 3
27	14	M1-Lead (PTT) output; site 4
28	39	M2-Lead (EVCD/Code Indicate) output; site 4
29	15	M1-Lead (PTT) output; site 5
30	40	M2-Lead (EVCD/Code Indicate) output; site 5
31	16	M1-Lead (PTT) output; site 6

**SDA Edge Connector Pinout***Table 4-4 SDA Edge Connector Pinouts (continued)*

<b>SDA Edge Connector Pin No.</b>	<b>Punchblock Pin No.</b>	<b>Signal Description</b>
32	41	M2-Lead (EVCD/Code Indicate) output; site 6
33	17	M1-Lead (PTT) output; site 7
34	42	M2-Lead (EVCD/Code Indicate) output; site 7
35	18	M1-Lead (PTT) output; site 8
36	43	M2-Lead (EVCD/Code Indicate) output; site 8
37	19	M1-Lead (PTT) output; site 9
38	44	M2-Lead (EVCD/Code Indicate) output; site 9
39	20	M1-Lead (PTT) output; site 10
40	45	M2-Lead (EVCD/Code Indicate) output; site 10
41	21	Audio in (ring)
42	46	Audio in (tip)
43	22	M1-Lead (PTT) input
44	47	M2-Lead (EVCD/Code Indicate) input
45	23	M1-Lead (PTT) input
46	48	M2-Lead (EVCD/Code Indicate) input
47	24	M1-Lead (PTT) cascable output
48	49	M2-Lead (EVCD/Code Indicate) cascable output
49	no connection	-48V supply in
50	25, 50	GND

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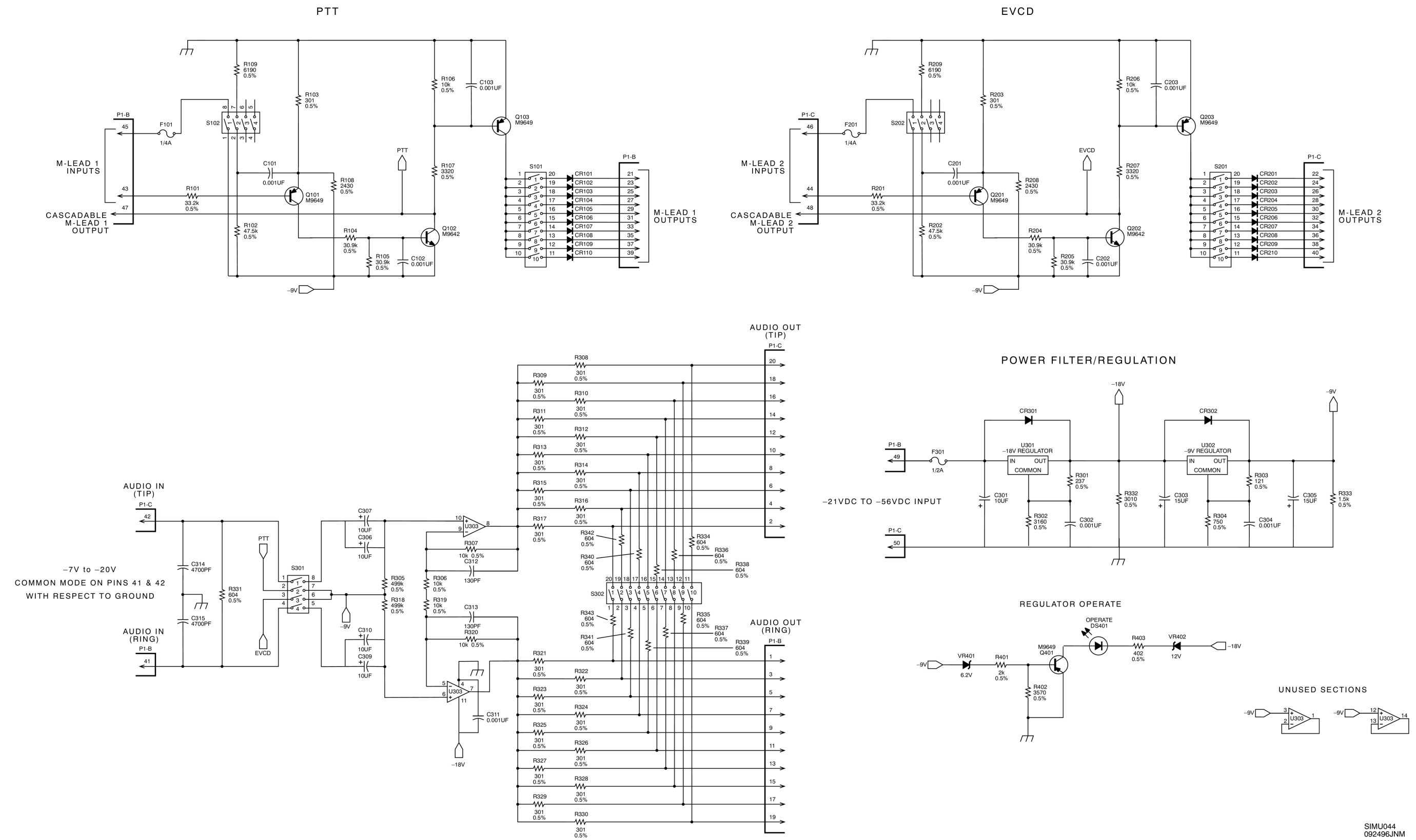
## SDA Parts List/Schematic Diagram

Figure 4-2 is the schematic diagram for the SDA. Refer to the parts list that follows for parts identification and ordering.

**SDA Parts List/Schematic Diagram**

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**Figure 4-2 SDA Schematic Diagram**

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## Simulcast Distribution Amplifier (QRN4878A) Parts List

## Simulcast Distribution Amplifier (QRN4878A) Parts List

Reference	Part Number	Description
<b>capacitor, fixed</b>		
		uF $\pm 10\%$ ; 100V unless otherwise stated
C101 thru 103	2111015D13	.001
C201 thru 203	2111015D13	.001
C301	2311019A24	10 $\pm 20\%$
C302	2111015D13	.001
C303	2311054H10	15; 25V
C304	2111015D13	.001
C305	2311054H10	15; 25V
C306, 307	2311054J12	10 $\pm 20\%$ ; 35V
C309, 310	2311054J12	10 $\pm 20\%$ ; 35V
C311	2111015D13	.001
<b>diode (see note)</b>		
CR101 thru 110	4884616A01	hot carrier
CR201 thru 210	4884616A01	hot carrier
CR301, 302	4882466H03	silicon
<b>light emitting diode (see note)</b>		
DS401	4884566R03	green
<b>fuse</b>		
F101, 201	6582408R04	1/4A 125V
F301	6582408R05	1/2A 125V
<b>transistor (see note)</b>		
Q101	4800869649	PNP
Q102	4800869642	NPN
Q103, 201	4800869649	PNP
Q202	4800869642	NPN
Q203, 401	4800869649	PNP
<b>resistor, fixed <math>\pm 0.5\%</math>; 1/4w</b>		
R101	0611040D42	33.2K
R102	0611040D57	47.5K
R103	0611040B44	301
R104, 105	0611040D39	30.9K
R106	0611040C91	10K
R107	0611040C45	3320
R108	0611040C32	2430
R109	0611040C71	6190
R201	0611040D42	33.2K
R202	0611040D57	47.5K
R203	0611040B44	301
R204, 205	0611040D39	30.9K
R206	0611040C91	10K
R207	0611040C45	3320
R208	0611040C32	2430
R209	0611040C71	6190
R301	0611040B34	237
R302	0611040C43	3160
R303	0611040B06	121

Reference	Part Number	Description
R304	0611040B82	750
R305	0611040E56	499K
R306, 307	0611040C91	10K
R308 thru 317	0611040B44	301
R318	0611040E56	499K
R319, 320	0611040C91	10K
R321 thru 330	0611040B44	301
R331	0611040B73	604
R332	0611040C41	3010
R333	0611040C12	1500
R334 thru 343	0611040B73	604
R401	0611040C24	2000
R402	0611040C48	3570
R403	0611040B56	402
<b>switch, rocker</b>		
S101	4083849F10	10-position
S102	4083849F09	4-position
S201	4083849F10	10-position
S202, 301	4083849F09	4-position
S302	4083849F10	10-position
<b>integrated circuit (see note)</b>		
U301, 302	5182276R36	Linear Voltage Regulator
U303	5182276R48	LW Quad Operational Amplifier
<b>voltage regulator (see note)</b>		
VR401	4883696E07	Zener, 6.2V
VR402	4882256C25	Zener, 12V
<b>non-referenced items</b>		
	4583716R01	CAM EJECTOR
	0982425R01	SOCKET, fuse (3 used)
	0210971A16	NUT, hex: M3x0.5 (2 used for U301)
	0310907C23	SCREW, machine: M3x0.5x10 (2 used for U301)
	0484152B01	WASHER, shoulder (2 used for U301)
	1483820M02	INSULATOR, heat conductive (U301)
	1484264C02	INSULATOR, rectangular (U301)
	2683662R02	HEAT SINK (U301)
	0210971A16	NUT, hex: M3x0.5 (2 used for U302)
	0310907C23	SCREW, machine: M3x0.5x10 (2 used for U302)
	0484152B01	WASHER, shoulder (2 used for U302)
	1483820M02	INSULATOR, heat conductive (U302)
	1484264C02	INSULATOR, rectangular (U302)
	2683662R02	HEAT SINK (U302)

**NOTE:** For optimum performance, diodes, transistors, crystals and integrated circuits must be ordered by Motorola part numbers.

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**Simulcast Distribution Amplifier (QRN4878A) Parts List**

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# ***DSM-II Module***

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## **Overview**

This chapter provides theory of operation and installation instructions for the Digital Simulcast Modem (DSM-II) module. The following topics are included in this chapter:

<b>Section</b>	<b>Page</b>	<b>Description</b>
Description	5-2	Describes the function and interfaces of the DSM-II module.
DSM-II Specifications	5-5	Lists the DSM-II specifications.
Description Of DSM-II Network Configurations	5-6	Describes the various modes and configurations that are used with the DSM-II module within the analog mode system.
DSM-II Card Programming Instructions	5-18	Provides general instructions on using the DSM-II programming functions.
DSM-II Module Installation	5-29	Provides instructions on installing the DSM-II module.
DSM-II Configuration Programming Instructions	5-31	Provides specific instructions on programming the DSM-II module for various configurations.
DSM-II Troubleshooting Instructions From the DSM-II Test Window	5-50	Describes the DSM-II test utilities and how to use the utilities.
DSM-II Phasing Using 1 PPS (GPS Operation)	5-63	Describes the GPS phasing operation used by DSM-II modules within a network.
DSM-II Pinouts	5-67	Lists the telco connector pinout for the DSM-II module.

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**Description**

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## Description

The DSM-II module is a user card that contains four full-duplex ports. It is installed in the TeNSr 800, 600, or 300 channel bank enclosure. (Refer to *TeNSr Manual*, for information on the TeNSr channel bank. See Chapter 1, Recommended Reference Publications, page 1-2.)

The DSM-II performs the amplitude and phase adjustment required for simulcast optimization in an analog (Non - ASTRO<sup>®</sup>) simulcast system in both the clear and SECURENET Four-Level Recovery Encode/Decode (FRED) modes. In the SECURENET<sup>®</sup> mode, the DSM-II grey-encodes and converts the Two-Level (2L) 12 kbit data to Four-Level (4L) 12 kbit data. In the SECURENET mode, it also distributes the transmit signal to all remote sites. In this manner, the DSM-II alone performs the functions of the FRED Prime, Simulcast Distribution Amplifier (SDA), and the Remote Delay Module (RDM) used in other systems. If necessary, the DSM-II can operate in the 2L mode. In addition to the clear audio and 12 kbit data, the DSM-II transmits High Speed Data (Trunking Systems), Low Speed Data (Trunking Systems), and Private Line/Digital Private Line (PL/DPL) (Conventional Systems).

## DSM-II Interface Function

In a simulcast system using T1/E1 distribution, the DSM-II is the core of the analog audio/12 kbit data network. The DSM-II routes all receive audio/12 kbit data (E&M signaling) from the remote site back to the comparator at the prime site. On the outbound side, the DSM-II distributes the transmit audio/12 kbit data out to the remote sites for transmission.

<b>NOTE</b>
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For 12 kbit operation, a SECURENET daughterboard is required.

ASTRO signals are never transported through the DSM-II card. Within a mixed-mode simulcast system, ASTRO signals travel over a separate digital link.

## DSM-II Signal Interfaces and Processing

### T1/E1 Interface

The DSM-II resides in the TeNSr channel bank. The TeNSr channel bank can be interfaced to either T1 or E1 communication channel links.

## Duplex Ports

The DSM-II is equipped with four full duplex ports per card. Each port is calibrated in terms of audio levels and the unique calibration coefficients are stored on the card. Therefore, for a given transmit and receive Transmission Level Point (TLP) setting and a given input level, the output level is the same for every DSM-II card. This allows the field-replacement of DSM-II cards without having to re-adjust the levels. The ports interface from the module to the system via a 50-pin female telco-type connector located on each DSM-II card.

## E-Lead Outputs

The DSM-II furnishes its E-lead output as normally open and normally closed relay contact closures. The relay contacts are interfaced through the telco-type connector located on each DSM-II card.

## Transport Time Delay Measurement and 1 Pulse-Per Second (PPS) Signal Distribution

The DSM-II performs real-time measurement of the system transport time delay per individual DSM-II port by the use of GPS time transfer techniques. The DSM-II receives the 1 PPS GPS signal from a local GPS receiver.

The DSM-II is capable of using only one source of 1 PPS from the GPS receiver per channel bank enclosure. Each enclosure can hold up to eight DSM-II cards, thereby providing 32 DSM-II ports per enclosure. Each DSM-II card has a 1 PPS input port (female BNC, 50 $\Omega$ , TTL level). The 1 PPS source can be connected to any one of the DSM-II cards in the enclosure. The DSM-II card that receives the 1 PPS signal distributes the signal to other DSM-II cards in the enclosure via the enclosure backplane. (If the card that has the 1 PPS connected to it must be removed, the 1 PPS source can be moved to any of the other DSM-II cards in the enclosure; the simulcast system delay will not change.)

It is possible to provide a 1 PPS source to each DSM-II card; however, only one input is required in normal operation. (It is recommended that more than only one card per chassis have a 1 PPS signal connected.) If that 1 PPS source fails, a good 1 PPS signal will automatically be selected from the other 1 PPS sources. If the DSM-II hardware used is Rev C or higher, then a single 1 PPS source can supply an entire site. On the Rev C hardware or higher, the input impedance for the 1 PPS can be jumpered for 50 $\Omega$  or high input impedance. When supplying multiple enclosures with a single 1 PPS source, one of the DSM-II cards is jumpered for 50 $\Omega$  (jumper in) and the others are configured for high-input impedance (jumper out). Refer to "GPS Input (1 PPS Input) Impedance Setting," page 5-64, for properly setting the impedance jumper.

## Pre-Programmed Time Delay Operation

The DSM-II is capable of automatically maintaining a preprogrammed time delay, independent of the actual communication path taken from the simulcast prime site to the simulcast remote site. This allows the DSM-II to be routed through Digital Crossconnect Switches (DCS) and through paths that have inconsistent

**Description**

delays (such as alternate path routing schemes). The delay amount is programmed through the Main Screen of the DSM-II card using a laptop Personal Computer (PC) and an emulator program.

**NOTE**

This preprogrammed time delay must be greater than the greatest time delay experienced or expected by the transport system.

**Remote Setup/Monitor Operation**

The DSM-II can be remotely provisioned (set up) through the built-in channel bank provisioning menu screens, or through the optional network management system.

Similarly, a remote-located DSM-II can be monitored for operational status from a local terminal.

**T1/E1 Error Detection**

The DSM-II is capable of detecting T1 or E1 errors using a CRC scheme. In this manner, any time stamp value that is invalid due to a transmission error is flagged and is not considered by the system as valid time stamp data.

**12 Kbit Communication**

In addition to SECURENET simulcast operation, the DSM-II is capable of transporting 12 kbit data from one point to another when a SECURENET daughterboard (Model 864) is installed on the DSM-II card.



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## DSM-II Specifications

The DSM-II was designed to match the same performance specifications as the TeNSr channel bank. Table 5-1 lists the channel bank specifications.

Table 5-1 **DSM-II Specifications**

Description	Value or Range
Operating temperature range	32° F to 122° F (0° C to 50° C)
Humidity	0% to 95%, noncondensing
M1-lead input	Capable of source/sink to ground (NOTE)
M2-lead input	Capable of source/sink to ground (NOTE)
E1-lead output	Normally open and normally closed relay contact outputs
E2-lead output	Normally open and normally closed relay contact outputs
Tx/Rx line interface	4-wire, 600 $\Omega$ balanced. Rear-panel RJ-27X (25-pair telco) connector.
M-lead current, maximum	3.6 mA
E-lead current, maximum	35 mA
<b>NOTE:</b> When M-lead is configured to sink ground, 10 V ( approximate) appears at input pin. When M-lead is configured to source ground, -10 V ( approximate) appears at input pin.	

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## Description Of DSM-II Network Configurations

The following three different network configurations using DSM-II cards are used in a simulcast system:

- ☐ Non-broadcast
- ☐ Broadcast
- ☐ Extended broadcast

The selected configuration for a particular link is dictated by the type of communications (clear or secure) over the link, and the number of sites and channels in the system.

For each configuration, the following paragraphs describe the arrangement of DSM-II cards throughout the system and the required programming of each card to facilitate the configuration.

### DSM-II Port Function Programming

Depending on the position of a particular DSM-II port within the system and the network configuration being set up, individual DSM-II ports are programmed to function as follows:

- ☐ Prime
- ☐ Remote
- ☐ Slave
- ☐ Loopback
- ☐ Slave-loopback

The choices of port function allow the DSM-II card to assume various roles within the system, depending on the position within the system of a particular DSM-II card. Individual DSM-II ports are programmed using a laptop PC which interfaces with the channel bank. (Instructions on setting up and loading programming using the laptop PC are discussed later in this chapter.)

### Non-Broadcast Mode Configuration

Figure 5-1 shows a three-site (two remote sites and one co-located site), single channel simulcast system configured for the non-broadcast mode. The Tx audio and E&M Lead signaling are distributed to the remote sites by a link consisting of DSM-II ports at the prime and remote sites, in conjunction with a Simulcast Distribution Amplifier (SDA) at the prime site. For every channel at each remote site, a DSM-II link (which consists of a DSM-II port at the prime site and DSM-II port at the remote site) is required.

When the DSM-II is configured for the non-broadcast mode, the applicable site programming field options are:

- ☐ **prime**
- ☐ **rem** (remote)
- ☐ **lb** (loopback)

Figure 5-1 shows the required port programming for the example non-broadcast set up shown. In Figure 5-1, the DSM-II port at the prime site is programmed as **prime** and the DSM-II port at the remote site is programmed as **rem**. For the co-located site, only one DSM-II port is required and it is programmed as **lb**.

Wide-Area Network (WAN) cards are used to interface the DSM-II modulated voice/12 kbit data onto the channel bank T1 or E1 links.

The non-broadcast mode is supported by V1.3 (or higher) DSM-II software and Rev A (or higher) DSM-II hardware releases.

<b>NOTE</b>
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If the channel bank covering the co-located site has a DSM-II only and the site parameter is set to <b>lb</b> , then a WAN card will still be required.
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## Description Of DSM-II Network Configurations

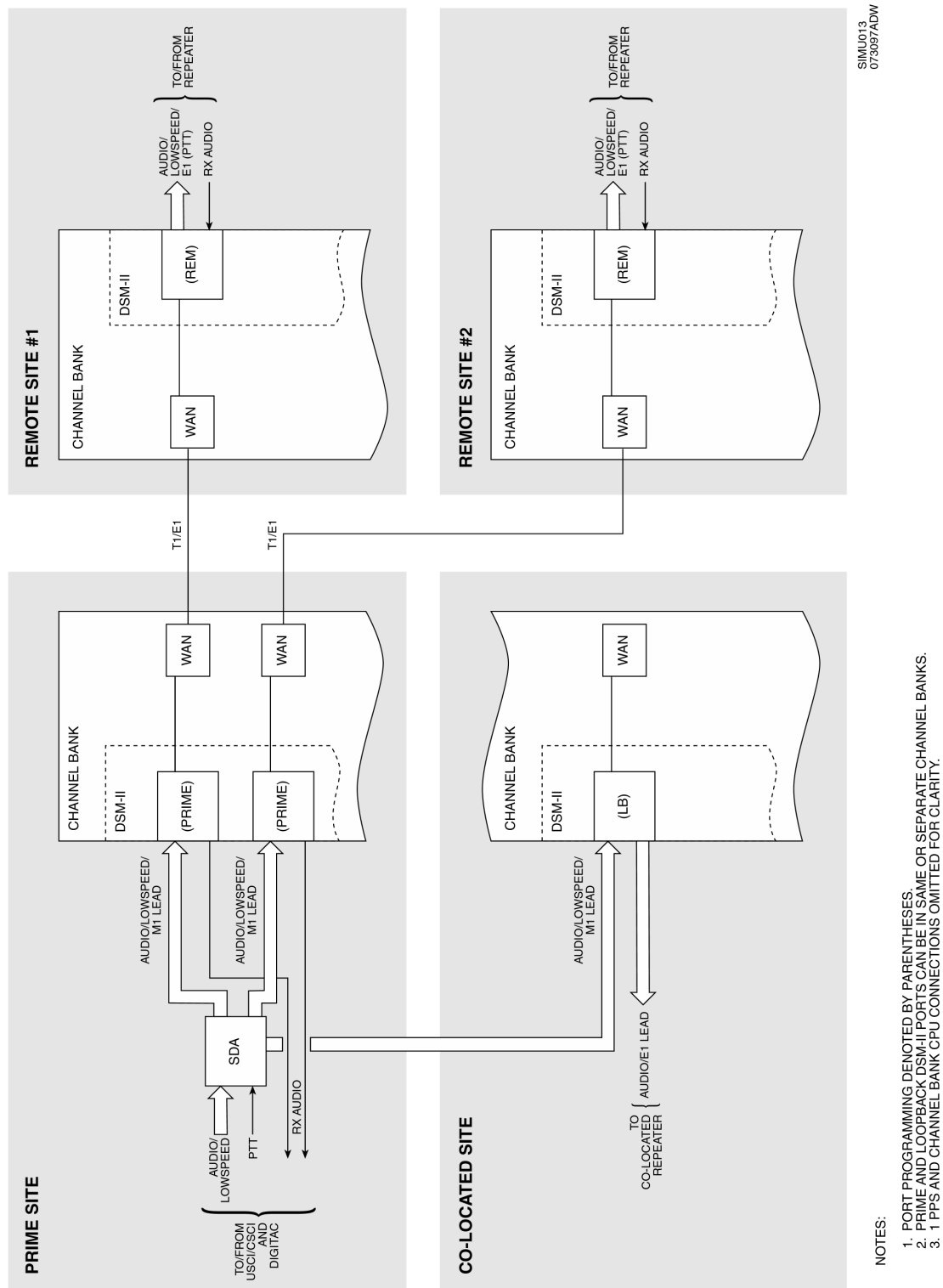


Figure 5-1 Non-broadcast Configuration Block Diagram

## Broadcast Mode Configuration

Figure 5-2 shows a three-site (two remote sites and one co-located site), single channel simulcast system configured for the broadcast mode. In the broadcast mode, the SDA is not used. The distribution function is performed internal to the channel bank at the WAN level.

When the DSM-II is configured for the broadcast mode, the applicable site programming field options are:

- ☐ **prime**
- ☐ **rem** (remote)
- ☐ **slave**
- ☐ **sl-lb** (slave-loopback)

Figure 5-2 shows the required port programming for the example broadcast setup shown. In Figure 5-2 for channels configured for the broadcast mode, only one port is programmed as **prime**. This port talks directly to a corresponding remote site DSM-II port which is programmed as **rem**. For each additional remote site, a DSM-II port is required at the prime site and the remote site. For these additional remote sites, the DSM-II port at the prime site is programmed as **slave** and the DSM-II port at the remote site is programmed as **rem**.

For a co-located site in the broadcast configuration, there is only one DSM-II port required and it is programmed as **sl-lb**. In order to configure a port as a slave or sl-lb, there must first be a valid source. Therefore, the prime port must be programmed first. (The slave and sl-lb get their Tx audio/12 kbit data by copying the transmit side of the prime port's DS0 time slot.)

The **SRC WAN** (source WAN) and **SRC TS** (source time slot) fields identify the source WAN and TS for the transmit audio for the slave or sl-lb ports. For a given simulcast channel on all slave and sl-lb ports, the **SRC WAN** and **SRC TS** are set to match the WAN identification (for example, W1-1 meaning port 1 of WAN card W1) and TS number of the prime port.

When a DSM-II is configured for the broadcast mode, it can support SECURENET operation when a SECURENET daughterboard is installed on every DSM-II card associated with the channel. With a SECURENET daughterboard present, the channel can be configured for either 2L or 4L operation by programming the mode for each port associated with the channel as either 2L or 4L. When the DSM-II is used in a SECURENET application, the M2-Lead (code detect) is required to place the DSM-II in the SECURENET mode of operation.

### NOTE

When a port is programmed as a slave or sl-lb, the transmit side of the port is non-functional. Any injected audio or M-Lead activation (through the physical interface or through the Test Screen) is ignored.

**Description Of DSM–II Network Configurations**

The broadcast mode configuration (with Secure) requires the hardware and firmware listed in Table 5-2:

*Table 5-2   **Hardware and Firmware for Extended Broadcast Mode with Secure***

<b>Description</b>	<b>Version/Rev Level</b>
DSM–II firmware	V2.4 or higher
DSM–II hardware	Rev C or higher
CPU firmware	V3.43 or higher

## Description Of DSM-II Network Configurations

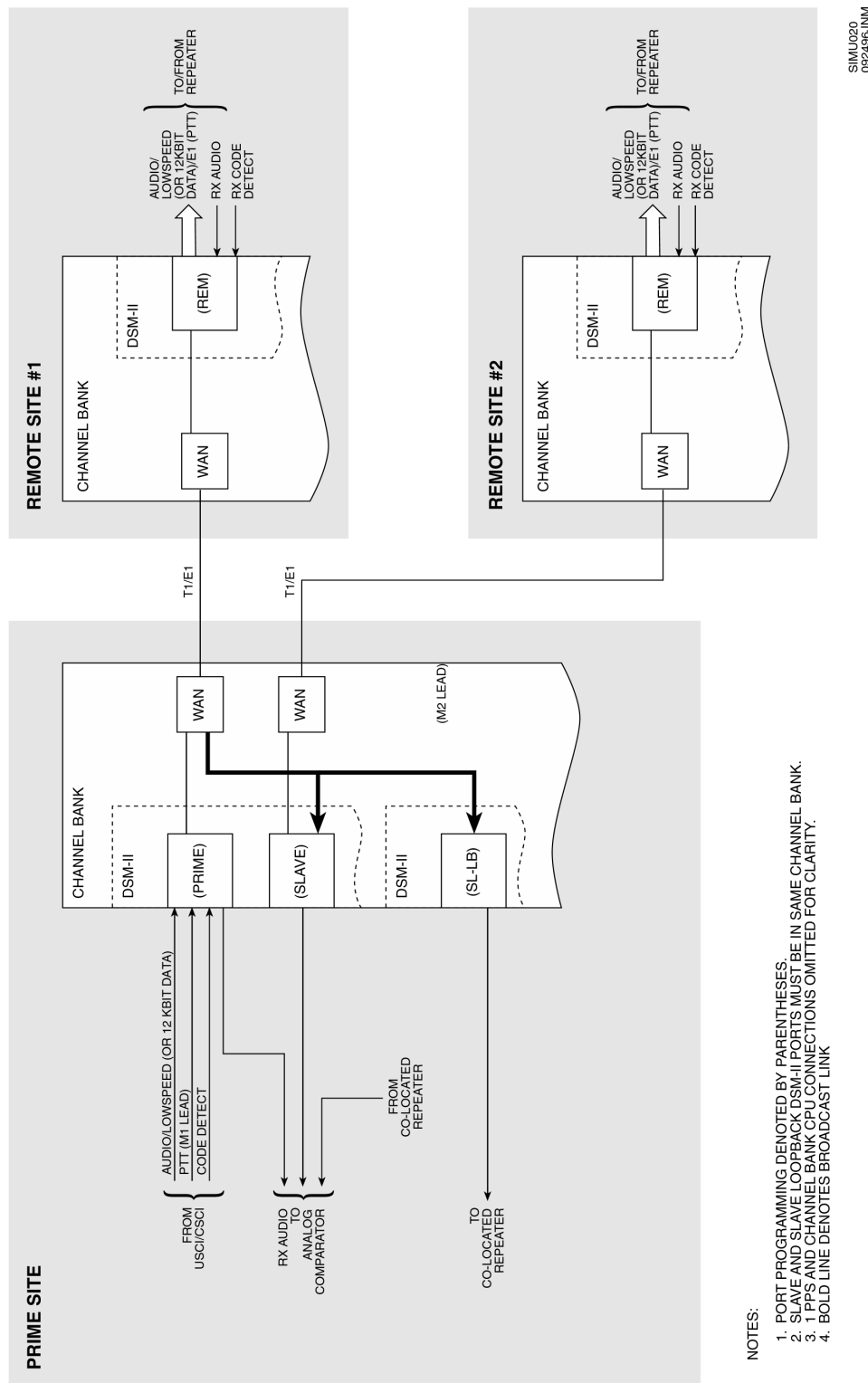


Figure 5-2 Broadcast Configuration Block Diagram

## Extended Broadcast Configuration

The extended broadcast configuration operates using the same principle as the broadcast configuration, but is used when a single TeNSr channel bank does not have the capacity to hold the required number of DSM-II cards at the prime site. In this case, additional channel banks are required at the prime site; the additional channel banks are referred to as sub-channel banks.

The need for extended broadcast is determined by multiplying the number of sites by the number of channels. If the product of the number of sites times the number of channels is greater than 28, then the extended broadcast configuration is required.

### NOTE

If the system has more than eight sites, then the extended broadcast configuration is required regardless of the number of voice channels.

The factor of 28 is derived as follows: In general, a simulcast system will use a Sub-Rate Unit (SRU) card to route the ReSC data link and the link to communicate with the remote site channel bank back to the prime site. The SRU uses one user slot in the TeNSr 800 chassis. This leaves seven user slots for seven DSM-II cards (or 28 DSM-II ports).

Figure 5-3 shows a four-site (three remote sites and co-located site) single-channel simulcast system using extended broadcast; this arrangement is used where 14 or fewer channels are required. Figure 5-4 shows a five-site (four remote sites and one co-located site) single channel simulcast system using the extended broadcast; the arrangement shown in Figure 5-4 is used where more than 14 channels are required.

### NOTE

A sub-channel bank generally is used on a per-site basis as follows: For systems with 14 or fewer voice channels, a sub-channel bank can cover two remote sites. (A TeNSr 800 chassis has eight user slots. In general, one of these user slots is used for an SRU card. This will leave seven user slots or 28 DSM-II card slots available. When the number of channels in the system is 14 or less, a single sub-channel bank is able to cover at least two remote sites. When the number of voice channels goes beyond 14, a single sub-channel bank will only be able to cover one remote site.)



When the DSM-II is configured for the extended broadcast mode, the applicable site programming field options are:

- ☐ **prime**
- ☐ **rem** (remote)
- ☐ **slave**
- ☐ **sl-lb** (slave-loopback)

### Extended Broadcast Configuration (14 or Fewer Channels)

Figure 5-3 shows the required port programming for an example extended broadcast setup with 14 or fewer channels. In the broadcast configuration, the prime and slave ports are located in the same channel bank for a given channel. However, in the extended broadcast configuration, the prime port and slave ports used for the extended links are in separate channel banks. The channel bank that contains the prime port is referred to as the master channel bank. Channel banks that contain slave ports are referred to as sub-channel banks. When a system has 14 or fewer channels, single channel bank can cover more than one remote site. For the second site in the master channel bank, the ports can be either a slave or sl-lb ports for a remote or co-located remote site, respectively.

In Figure 5-3, the prime port talks directly (via W1-1) to a DSM-II port at remote site 1. Once this connection has been made, a slave port in the master channel bank is broadcasted into W1-2 which, in turn, connects to remote site 2. (In place of the master channel bank slave port shown in Figure 5-3, an sl-lb port can be substituted to directly connect to a co-located remote, if desired.) The prime port connection to W1-1 also broadcasts into W2-1 and W2-2. These WANs (W2-1 and W2-2) are in turn connected to WAN W1-1 in sub-channel banks 1 and 2, respectively.

Sub-channel bank 1 has an E1/T1 connection (W1-2) out to a remote site. Within a sub-channel bank for a given channel and remote site, there is a DSM-II port that is programmed as a slave. The slave is assigned to a time slot in W1-2. The source audio will come from W1-1 (W1-1 is the E1/T1 link that contains the broadcast of the prime port).

In Figure 5-3, sub-channel bank 2 is the co-located channel bank. For each channel, a DSM-II port is required in the co-located channel bank. Unlike the non-broadcast (DSM-II port programmed as **lb**) and the broadcast mode (DSM-II port programmed as **sl-lb**), the DSM-II port for the co-located site in the extended broadcast mode is programmed as a **rem**. Since the DSM-II port is programmed as a **rem**, there is no "SRC WAN" and "SRC TS;" the programmed WAN assignment is the WAN that accepts the broadcast link from the master channel bank. This link is a T1/E1 link that contains the broadcast of the prime port. Similarly, the timeslot assignment for the co-located remote DSM-II is the matching time slot that the prime port was broadcast into.

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Description Of DSM-II Network Configurations

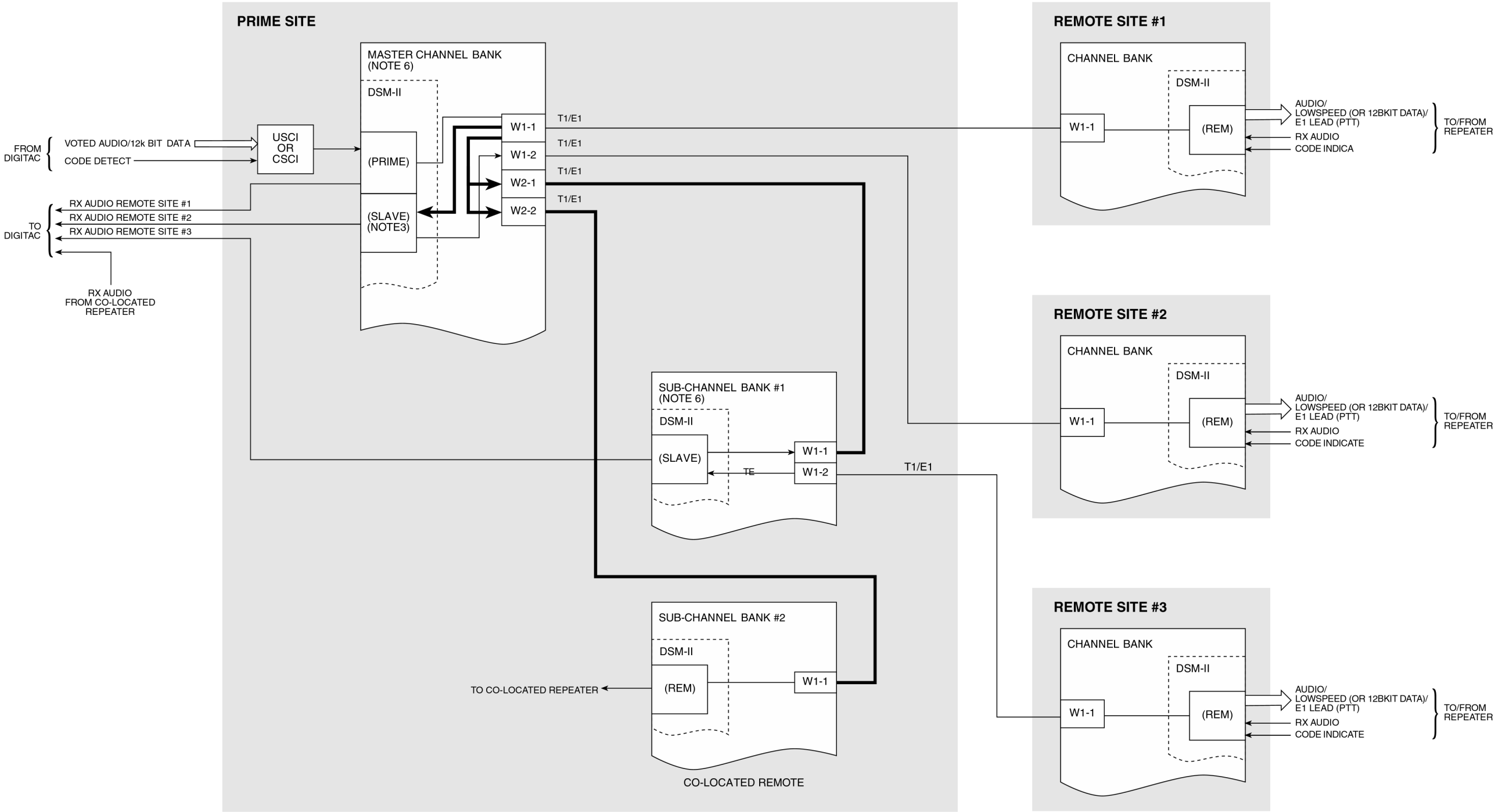
<b>NOTE</b>
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When the port for the co-located site is in the master channel bank, it is configured as a **sl-lb**.

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### **Extended Broadcast Configuration (Greater than 14 Channels)**

Figure 5-4 shows the required port programming for an example extended broadcast setup with more than 14 channels. In the broadcast configuration, the prime and slave ports are located in the same channel bank for a given channel. However, in the extended broadcast configuration, the prime port and slave ports are in separate channel banks. The channel bank that contains the prime port is referred to as the master channel bank. Channel banks that contain slave ports are referred to as sub-channel banks.



- NOTES:
1. PORT PROGRAMMING DENOTED BY PARENTHESES.
  2. 1 PPS AND CHANNEL BANK CPU CONNECTIONS OMITTED FOR CLARITY.
  3. MASTER OR SUB-CHANNEL BANK CAN DIRECTLY SUPPORT CONNECTION TO REMOTE SITE #1 AND ADDITIONALLY SUPPORT CONNECTION TO SL-LB CO-LOCATED REMOTE OR SECOND REMOTE. (SECOND REMOTE CONNECTION SHOWN HERE.)
  4. MASTER AND ANY SUB-CHANNEL BANK CAN COVER TWO SITES PER BANK (14 OR LESS CHANNEL CONFIGURATION ONLY).
  5. BOLD LINES DENOTE BROADCAST LINKS.
  6. REQUIRES V4.X OR HIGHER CPU FIRMWARE.

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Figure 5-3    **Extended Broadcast Configuration Block Diagram (14 or fewer channels)**

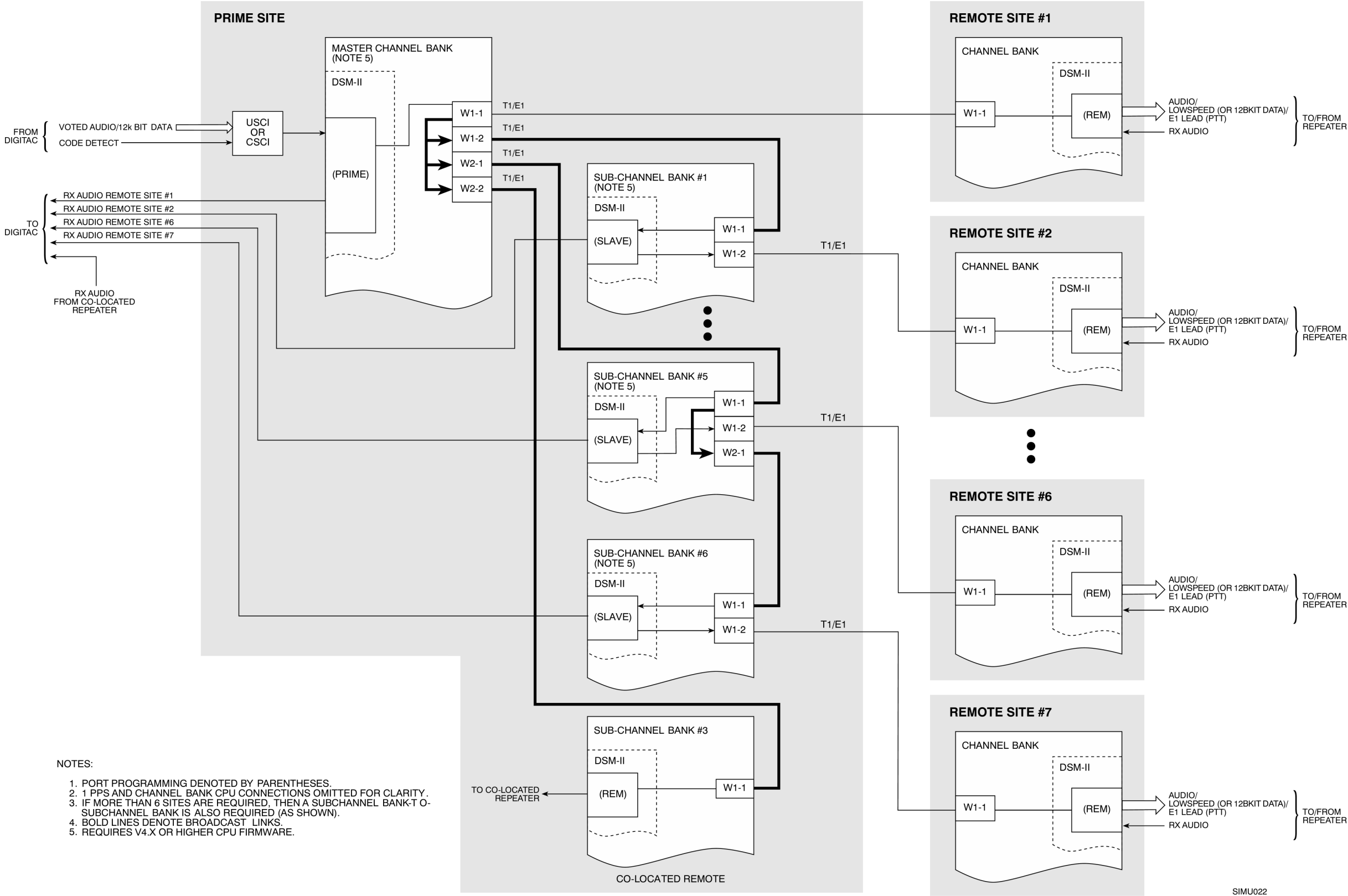


Figure 5-4    **Extended Broadcast Configuration Block Diagram (14+ channels)**

In Figure 5-3, the prime port talks directly (via W1-1) to a DSM-II port At the remote site 1. Once this connection has been made, the prime port is broadcasted into W1-2, W2-1, and W2-2. These WANs (W1-2, W2-1, and W2-2) are in turn connected to WAN W1-1 in sub-channel bank 1, 5, and 3, respectively.

Sub-channel banks 1 and 5 have an E1/T1 connection (W1-2) out to a remote site. Within these channel banks for a given channel, there is a DSM-II port which is programmed as a slave. The slave is assigned to a time slot in W1-2. The source audio will come from W1-1 (W1-1 is the E1/T1 link that contains the broadcast of the prime port).

In Figure 5-4, sub-channel bank 3 is the co-located channel bank. For each channel, a DSM-II port is required in the co-located channel bank. Unlike the non-broadcast (DSM-II port programmed as **lb**) and the broadcast mode (DSM-II port programmed as **sl-lb**), the DSM-II port for the co-located site in the extended broadcast mode is programmed as a **rem**. Since the DSM-II port is programmed as a **rem**, there is no SRC WAN and SRC TS; the programmed WAN assignment is the WAN that accepts the broadcast link from the master channel bank. This link is the E1/T1 link that contains the broadcast of the prime port. Similarly, the timeslot assignment for the co-located remote DSM-II is the matching time slot that the prime port was broadcast into.

Sub-channel bank 6 is the same as channel banks 1 and 5, except that the broadcast WAN will come from channel bank 5 via WAN W2-1. A TeNSr 800 chassis is limited to eight WANs. In a simulcast system using extended broadcast mode, the master channel bank should be equipped with the redundant WAN option. This will leave six WANs routing T1/E1 channels out to sites as well as to sub-channel channel banks. When the number of voice channels in a system is greater than 14 and there are more than six remote sites, the broadcast link for the sub-channel that covers the seventh remote site will have to come from another sub-channel bank. In this case, the broadcast WAN for the sixth sub-channel bank will have to come from a sub-channel bank as shown in Figure 5-4 by the link between sub channel bank 5, W2-1 and sub channel bank 6, W1-1.

The extended broadcast mode configuration requires the hardware and firmware listed in Table 5-3:

Table 5-3 **Hardware and Firmware for Extended Broadcast Mode**

Description	Version/Rev Level
DSM-II firmware	V2.4 or higher
DSM-II hardware	Rev C or higher
CPU firmware (master channel bank)	V4.X or higher
CPU firmware (sub-channel bank with slave ports)	V4.X or higher
CPU firmware (sub-channel bank with remote ports)	V3.43 or higher
CPU firmware (remote channel bank)	V3.43 or higher

**DSM-II Card Programming Instructions**

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## **DSM-II Card Programming Instructions**

The functions of individual DSM-II ports are set up using the laptop PC and the emulator software (PCPLUS, or equivalent). In using the PC and software, two menu windows are used to set up the DSM-II cards. These windows are:

- ☐ Channel bank Main Window – used to access a particular DSM-II card within a channel bank
- ☐ DSM-II Card Window – used to set up the individual ports of a DSM-II card once the card has been selected from the Main Window

The following sections describe how to access DSM-II cards from the Main Window, and how to set up a DSM-II card from the DSM-II Card Window.

### **Accessing a Particular DSM-II Card From The Main Window**

1. Connect the laptop PC serial port to the TERM connector on channel bank interface card.
2. Log on to the channel bank where DSM-II card is installed. The Main Window appears.
  - 2.1 Note the channel bank user slot location of the DSM-II card that is intended to be accessed.
3. From the Main Window, access the DSM-II card by moving the cursor to the user slot that corresponds to intended DSM-II card.
4. Press the **Enter** key. The DSM-II card is now selected. The DSM-II Card Window appears, as shown in Figure 5-5 and described in the following sections.

## DSM-II Card Window Fields Overview

Figure 5-5 shows the DSM-II Card Window. Table 5-4 lists the different options for the DSM-II Card Window fields and their default values.

When the DSM-II Card Window is accessed, the laptop PC displays several fields which, when set, control the function of a given DSM-II port. The following sections provide an overview of the function of each DSM-II Card Window field. Each section describes how to access and set the options of each field.

Fremont : U1 DSM-4 3164 Rev A0-0 Ser 0001 : 01-07-99 12:49				
Version #:1.1				
	1	2	3	4
STATE	actv	actv	actv	actv
SITE	rem	rem	rem	rem
WAN	w1-1	w1-1	w1-1	w1-1
TS	15	16	17	18
SRC WAN	n/a	n/a	n/a	n/a
SRC TS	n/a	n/a	n/a	n/a
MODE	clear	clear	clear	clear
Rx TLP CLR	0.0	0.0	0.0	0.0
Rx TLP SEC	-10.0	-10.0	-10.0	-10.0
Tx TLP CLR	0.0	0.0	0.0	0.0
M POLARITY	++	++	++	++
MAX DELAY	20000	20000	20000	20000
PRIME TYPE*	int	int	int	int
Save : Undo : Refresh : Copy : Test : Main				

\* Only appears with V4.X Firmware.

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Figure 5-5 **DSM-II Card Window**

**DSM-II Card Programming Instructions***Table 5-4 Default DSM-II Card Window Fields*

Field	User Options	Default
State	<ul style="list-style-type: none"> <li>standby</li> <li>active</li> </ul>	standby
Site	<ul style="list-style-type: none"> <li>rem (remote)</li> <li>prime</li> <li>slave</li> <li>lb</li> <li>sl-lb</li> </ul>	rem
WAN	<ul style="list-style-type: none"> <li>n/a</li> <li>w1-1</li> <li>w1-2</li> <li>w2-1</li> <li>w2-2</li> <li>w3-1</li> <li>w3-2</li> <li>w4-1</li> <li>w4-2</li> <li>None (V4.X default)</li> </ul>	w1-1
TS	<ul style="list-style-type: none"> <li>01-24 (T1)</li> <li>01-31 (E1)</li> </ul>	01
SRC WAN	<ul style="list-style-type: none"> <li>n/a</li> <li>w1-1</li> <li>w1-2</li> <li>w2-1</li> <li>w2-2</li> <li>w3-1</li> <li>w3-2</li> <li>w4-1</li> <li>w4-2</li> <li>None (V4.X default)</li> </ul>	n/a
SRC TS	<ul style="list-style-type: none"> <li>n/a</li> <li>01-24 (T1)</li> <li>01-31 (E1)</li> </ul>	n/a
Mode	<ul style="list-style-type: none"> <li>clear</li> <li>2L</li> <li>4L</li> <li>e&amp;m</li> </ul>	clear
Rx TLP CLR	–16.3 through 6.8	0.0
Rx TLP SEC	–12.0 through –4.0	–10.0



## DSM-II Card Programming Instructions

Table 5-4 *Default DSM-II Card Window Fields (continued)*

Field	User Options	Default
Tx TLP CLR	-16.3 through 6.8	0.0
M polarity	<ul style="list-style-type: none"> <li>• + +</li> <li>• - -</li> <li>• + -</li> <li>• - +</li> </ul>	++
Max delay	18000 through 65000	20000
Prime type (NOTE)	<ul style="list-style-type: none"> <li>• int</li> <li>• ext</li> <li>• n/a</li> </ul>	int
<b>NOTE:</b> Appears only with V4.X CPU firmware.		

## DSM-II Card Window Editing Options

At the bottom of the DSM-II Card Window, there are several generic editing options that are used in the course of programming DSM-II ports. Table 5-5 lists these options.

Whenever a configuration is developed, any changes in the course of developing the configuration should be saved immediately after entering the change.

Table 5-5 *DSM-II Card Window Editing Options*

Option	Function
<b>Save</b>	Saves changes to settings. When changes have been made in the DSM-II Card Window, an asterisk appears next to the port number where changes were made. This asterisk indicates that changes have been made but not saved. Press the <b>S</b> key to save changes.
<b>Undo</b>	Returns all settings to the last saved state. Press the <b>U</b> key to return to last saved state.
<b>Refresh</b>	(reserved for future use)
<b>Copy</b>	Copies the contents of the current port (column) to the next port (column). This option is useful when a lot of entries that are changed in one port are repeated in the subsequent ports. Press <b>C</b> to copy the current port settings to the next.
<b>Test</b>	The test option enters the Test Window of the DSM-II. Press the <b>T</b> key to enter the Test Window.
<b>Main</b>	Returns to the Main Window of the channel bank. Press the <b>M</b> key to return to the Main Window of the channel bank.

**DSM-II Card Programming Instructions****DSM-II Card Window Fields Detailed Descriptions**

Each of the DSM-II Card Window fields and their uses are described in the following sections. Instructions on how to access and set the fields are also provided.

**State**

The State field determines whether the port is active or inactive. Set this field as follows:

1. Using the arrow keys, highlight the STATE field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. Options for STATE field are now displayed at bottom of window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.

Set the State field to standby (stdby) for ports you are not using or have not yet configured. Set it to active (actv) for ports that are ready for use.

**Site**

The Site field determines the function of a particular DSM-II port. The options for the site field are:

- ☐ **rem**
- ☐ **prime**
- ☐ **slave**
- ☐ **lb**
- ☐ **sl-lb**

Set this field as follows:

1. Using the arrow keys, highlight the SITE field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. Options for SITE field are now displayed at bottom of window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.

Specific instructions for setting the Site field are provided in the “DSM-II Configuration Programming Instructions,” page 5-31.

## Wide-Area Network (WAN)

The WAN field identifies the WAN link assigned to a particular port. Depending on the number of T1s or E1s in the channel bank, there can be up to eight options for WAN assignment. The WAN field applies only to **rem**, **prime**, and **slave** ports.

Set the WAN field for a given port as follows:

1. From the DSM-II Card Window, highlight the WAN field for the intended port by using the arrow keys.
2. When the desired field is highlighted, press the **Enter** key. Options for WAN field are now displayed at bottom of window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.

### NOTE

Each WAN link listing consists of the WAN card number (first digit following “W”) and WAN port of the WAN card (second digit following “-”). For example, “W2-1” denotes port 1 of WAN 2.

Specific instructions for setting the WAN field are provided in the “DSM-II Configuration Programming Instructions,” located on page 5-31, as applicable.

## TS

The TS field identifies the time slot on the WAN link to which a particular port is assigned. Values range from 1-24 for T1 links and 1-31 for E1 links.

### NOTE

Time slot 16 is not available for E1 links that are programmed for CAS or CCS signaling.

The TS field applies only to **rem**, **prime**, and **slave** ports.

Set the TS field as follows:

1. From the DSM-II Card Window, position the cursor over the TS field corresponding to the desired port.
2. With the intended field highlighted, press the **Enter** key. Entry box for time slot number appears at bottom of window.

**DSM-II Card Programming Instructions**

3. Using the up and down arrow keys, toggle through time slot numbers until the desired number is displayed. Press the **Enter** key.

Specific instructions for setting the TS field are provided in the “DSM-II Configuration Programming Instructions,” page 5-31, as applicable.

**SRC (Source) WAN and SRC TS**

The SRC WAN and SRC TS fields identify the WAN link and time slot from where a slave and sl-lb port will get their transmit data. For a given channel, the SRC WAN and SRC TS setting is the WAN setting and TS setting of the prime port on the channel. For example, if the prime port WAN setting and TS setting on a given channel is W1-2, time slot 01, then the SRC WAN setting and time slot setting for all slave and sl-lb ports associated with that channel is W1-2, time slot 01.

In the extended broadcast configuration when the slave port’s PRIME TYPE parameter is set to “Ext,” the SRC WAN and SRC TS are set to the WAN and TS within the broadcast link. The broadcast link is a T1/E1 link from the master channel bank (or another sub-channel bank) that contains the broadcast of the prime port.

Depending on the number of T1s or E1s in the channel bank, there can be up to eight options for SRC WAN assignment. The SRC WAN field applies only to slave and sl-lb ports.

**Mode**

The Mode field selects the port encoding mode. The options are:

- ☐ Clear
- ☐ 2L (two-level Secure)
- ☐ 4L (four-level Secure, or FRED)
- ☐ E&M

**NOTE**

The E&M Mode selection is currently undefined.

When set to...	Port can operate in these modes...
clear	clear mode only
2L	clear and 2L SECURENET modes
4L	clear and 4L (FRED) SECURENET modes

Set this field as follows:

1. Using the arrow keys, highlight the MODE field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. Options for MODE field are now displayed at bottom of window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.

<b>NOTE</b>
-------------

The non-broadcast configuration supports clear only.

The broadcast and extended broadcast configurations support clear and SECURENET modes.

On a given channel, all ports associated with the channel must have the mode (clear, 2L, 4L) programmed the same.

When the DSM-II is programmed for SECURENET mode, it requires a SECURENET daughterboard to process the SECURENET data.

## Rx TLP CLR

The Receive Transmission Level Plan Clear field adjusts the audio levels at the 600 $\Omega$  output. The values (expressed in dB) for this field are -16.3 through 6.8. The Rx TLP CLR acts as an amplifier or attenuator. Entering a positive value will amplify the level on the receive side of the DSM-II port by the dB value entered. For example, a value of 2.0 dB will increase the output level by 2.0 dB. Entering a negative value will decrease the level on the receive side of the DSM-II port by the dB value entered. The default value of 0.0 dB will not increase or decrease the output value at the corresponding DSM-II port.

Set this field as follows:

1. Using the arrow keys, highlight the Rx TLP CLR field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. An entry box appears at the bottom of the window.
3. Enter the desired value. Press the **Enter** key when done.

**DSM-II Card Programming Instructions****NOTE**

When initially setting up ports, leave this field set to its default value.

**RX TLP SEC**

The Receive Transmission Level Plan SECURENET field adjusts the SECURENET data levels at the 600 $\Omega$  output. The values (expressed in dBm) for this field are -12.0 through -4.0. The Rx TLP SEC is an absolute value in dBm. The default value of -10 dBm will output a SECURENET level of -10 dBm.

Set this field as follows:

1. Using the arrow keys, highlight the Rx TLP SEC field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. An entry box appears at the bottom of the window.
3. Enter the desired value. Press the **Enter** key when done.

**NOTE**

When initially setting up ports, leave this field set to its default value.

**TX TLP CLR**

The Transmit Transmission Level Plan Clear field adjusts the audio levels at the 600 $\Omega$  input. The values (expressed in dB) for this field are -16.3 through 6.8. The Tx TLP CLR acts as an attenuator. Entering a positive value will attenuate the level on the receive side of the corresponding DSM-II port by the entered dB value. For example, a value of 2.0 dB will decrease the output level at the corresponding DSM-II port by 2.0 dB. Entering a negative value will increase the output level by the dB value entered. The default value of 0.0 dB will not increase or decrease the output value at the corresponding DSM-II port.

Set this field as follows:

1. Using the arrow keys, highlight the Tx TLP CLR field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. An entry box appears at the bottom of the window.
3. Enter the desired value. Press the **Enter** key when done.

**NOTE**

When initially setting up ports, leave this field set to its default value.

**M-Lead Polarity**

The M-Lead Polarity field sets the M-Leads to sink or source ground. The options for this field are:

- ☐ ++
- ☐ --
- ☐ + -
- ☐ - +

The first sign (+ or -) corresponds to the M1-Lead and the second sign (+ or -) corresponds to the M2-Lead.

When the M-Lead Polarity is programmed with a +, it is configured to sink ground. All Motorola equipment contains this type of interface for the M-Leads (with the exception of the SDA).

When the M-Lead Polarity is programmed with a -, it is configured to source ground. When interfacing the SDA to the DSM-II, the M-Leads are required to source ground.

Set this field as follows:

1. Using the arrow keys, highlight the M-LEAD POLARITY field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. Options for M POLARITY field are now displayed at the bottom of window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.

**MAX Delay**

The **MAX Delay** field adjusts the end-to-end delay (throughput delay) from the 600 $\Omega$  input at the prime site DSM-II port to the 600 $\Omega$  output at the remote site DSM-II. The end-to-end delay of the prime-to-remote path consists of three parts:

- ☐ The DSM-II card delay
- ☐ The network delay (T1 or E1)
- ☐ A buffer.

**DSM-II Card Programming Instructions**

This cumulative delay is described further in the “DSM-II Phasing Using 1 PPS (GPS Operation), page 5-63 section. The MAX Delay has no effect on the throughput delay from the remote site to the prime site; the throughput delay from the remote site to the prime site is fixed at approximately 15 ms.

Set this field as follows:

1. Using the arrow keys, highlight the MAX DELAY field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. An entry box appears at the bottom of the window.
3. Enter the desired value. Press the **Enter** key when done.

**NOTE**

When initially setting up ports, leave this field set to its default value.

**Prime Type**

The Prime Type field applies only to the slave port in the extended broadcast configuration. The options for this field are **int** and **ext**.

In the extended broadcast configuration, since the prime port and slave ports may be in separate channel banks, this field specifies whether the slave is sourcing from a prime port that is external or internal to the channel bank containing the particular slave port.

**NOTE**

The prime type field will only appear when the CPU firmware is version is V4.X or higher.

Set this field as follows:

1. Using the arrow keys, highlight the Prime Type field corresponding to the desired port.
2. When the desired field is highlighted, press the **Enter** key. Options for PRIME TYPE field are now displayed at the bottom of the window.
3. Use the left and right arrow keys to toggle through the options. When the desired option is highlighted, press the **Enter** key.



---

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## DSM-II Module Installation

Install DSM-II module(s) and perform port setup programming as described in the following sections.

### Module Installation

Install DSM-II module(s) into channel bank user slots as follows:

<b>NOTE</b>
-------------

The DSM-II card can be hot-inserted.

1. Align DSM-II with user slot.
2. Insert and fully seat DSM-II into slot.
3. Secure DSM-II in slot using captive screw at each end of the module.
4. Connect appropriate duplex communication cable within rack to telco connector on DSM-II.
5. Set 1 PPS impedance jumper as required. Refer to “GPS Input (1 PPS Input) Impedance Setting,” page 5-64 for more information, if required.
6. Proceed to Port Programming and Set-up in the following section.

### Port Programming and Set-up

<b>NOTE</b>
-------------

DSM-II cards must be installed in all intended channel bank locations prior to port programming and set-up.

Connect laptop PC to channel bank where DSM-II is being installed as described in as described in the following section, “Connecting the Laptop PC to Channel Bank.” Depending on the port configuration required, proceed to the following sections, as applicable:

- ☐ “Non-Broadcast Setup,” page 5-31,
- ☐ “Broadcast Setup,” page 5-33
- ☐ “Extended Broadcast Setup,” page 5-36.

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**DSM-II Module Installation****Connecting the Laptop PC to Channel Bank**

A laptop PC with the appropriate emulator software (PCPLUS, or equivalent) is interfaced to the channel bank (and, in turn, the DSM-II cards) via an interface card within the channel bank.

Connect the laptop PC serial port to the TERM connector on the channel bank interface card.

---

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## DSM-II Configuration Programming Instructions

The following sections describe the network interfaces and programming used for setting up links in the non-broadcast, broadcast, and extended broadcast configurations. Since the network is extremely flexible in the amount and types of actual set-ups that can be installed, the following sections do not describe how to set up any specific, exact configuration. However, the following sections describe rules and common practices that are applicable to their corresponding configurations. Use the information in the following sections as a guide when setting up actual network links.

<b>NOTE</b>
-------------

Discussion and examples in the following sections are representative only and do not necessarily describe a specific configuration.

---

The following procedures describe setups for a given channel. Procedures must be repeated for all channels, as applicable.

---

### Non-Broadcast Setup

Perform non-broadcast configuration setup as follows:

1. At prime site, connect the laptop PC serial port to the TERM connector on the channel bank interface card.
2. Log on to the channel bank where prime port DSM-II cards are installed. The Main Window appears.
  - 2.1 Note the channel bank user slot location of the DSM-II that is intended to provide the prime port.
3. From the Main Window, access the DSM-II that corresponds to the intended DSM-II prime port.
4. Press the **Enter** key. The DSM-II card is now selected. The DSM-II Card Window now appears.

**DSM-II Configuration Programming Instructions**

5. Set up the fields, as applicable, (along with the corresponding settings) as listed in the following table for each DSM-II port.

Field	Setting
STATE	stdby
SITE	prime
MODE	clear
M POLARITY	Set as required. (See "M-Lead Polarity" section above for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default

6. The WAN and TS fields can now be set. For each DSM-II port, assign the desired WAN and time slot. Each prime port is now set to link with a remote port.
7. Set port to "actv."
8. From the Main Window, press the **S** key to save the setup.
9. At the remote site, connect the laptop PC to the appropriate channel bank as described in "Connecting the Laptop PC to Channel Bank," page 5-30.
10. Log on to the channel bank where remote port DSM-II cards are installed. The Main Window appears.
  - 10.1 Note the channel bank user slot location of the DSM-II card that is intended to serve as remote port.
11. From the Main Window, access the DSM-II card that corresponds to the intended DSM-II remote port.
12. Press the **Enter** key. The DSM-II card is now selected. The DSM-II Card Window now appears.
13. Set up the fields, along with the corresponding settings, listed in the following table for each DSM-II port, as applicable.

Field	Setting
STATE	stdby
SITE	rem
MODE	clear
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY	leave default

## DSM-II Configuration Programming Instructions

14. The WAN and TS fields can now be set. For each DSM-II port, assign the desired WAN and time slot.
15. Set port to “actv.”
16. From the Main Window, press the **S** key to save the setup.
17. Set up co-located remote (if used) as follows:
  - 17.1 Set up the fields (along with the corresponding settings) listed in the following table for each DSM-II port, as applicable.

Field	Setting
STATE	stdby
SITE	lb
MODE	clear
M POLARITY	Set as required. (See “M Polarity” section above for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY	leave default

- 17.2 The WAN and TS fields can now be set. For each DSM-II port, assign the desired WAN and time slot.
- 17.3 Set port to “actv.”
- 17.4 From the Main Window, press the **S** key to save the setup.
18. Repeat steps 1 through 19.4 for any additional channels.

## Broadcast Setup

Perform broadcast configuration setup as follows:

1. At the prime site, connect the laptop PC serial port to the TERM connector on the channel bank interface card.
2. Log on to the channel bank where the prime site DSM-II card(s) is installed. The Main Window appears.
3. Set up DSM-II prime port as follows:
  - 3.1 Note the channel bank user slot location of the DSM-II card that is intended to provide the prime port.
  - 3.2 From the Main Window, access the DSM-II card that corresponds to intended DSM-II prime port.
  - 3.3 Press the **Enter** key. The DSM-II card is now selected. The DSM-II Card Window now appears.

**DSM-II Configuration Programming Instructions**

- 3.4 Set up the fields (along with the corresponding settings) listed in the following table for the DSM-II prime port.

Field	Setting
STATE	stdby
SITE	prime
MODE	Set as required. (See “Mode,” page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default

- 3.5 Assign the desired WAN and time slot to DSM-II prime port.
- 3.6 Set port to “actv.”
- 3.7 From the Main Window, press the **S** key to save the setup.
4. Set up each DSM-II slave and sl-lb port as follows:

<b>NOTE</b>
-------------

The first DSM-II prime port talks directly to a corresponding remote site DSM-II port which is programmed as **rem**. For each additional remote site, a DSM-II port is required at the prime site and the remote site. For these additional remote sites, the DSM-II port at the prime site is programmed as **slave** and the DSM-II port at the remote site is programmed as **rem**. For a co-located site, only one DSM-II port located at the prime site is required; this port is programmed as **sl-lb**.

- 4.1 From the Main Window, access the DSM-II card that corresponds to intended DSM-II slave port.
- 4.2 Press the **Enter** key. The DSM-II card is now selected. The DSM-II Card Window now appears.

## DSM-II Configuration Programming Instructions

- 4.3** Set up the fields (along with the corresponding settings) listed in the following table for DSM-II slave or sl-lb ports.

Field	Setting
STATE	stdby
SITE	NOTE 1
MODE	Set as required. (See "Mode," page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY (NOTE 2)	leave default
<p><b>NOTE:</b> 1. Set SITE field as follows:</p> <ul style="list-style-type: none"> <li>• For DSM-II port serving remote ports, set as "slave."</li> <li>• For DSM-II port serving co-located port, set as "sl-lb."</li> </ul> <p>2. MAX Delay field applies only to sl-lb port.</p>	

- 4.4** Assign the desired WAN and time slot to slave port.
- 4.5** Access the SRC WAN field for the slave or sl-lb port being set up. From the options, select the SRC WAN assignment that matches the WAN assigned to the prime port.
- 4.6** Access the SRC TS field for the slave or sl-lb port being set up. From the options, select the SRC TS assignment that matches the time slot assigned to the prime port.
- 4.7** Set the SITE field for slave and sl-lb ports to "actv."
- 4.8** From the Main Window, press the **S** key to save the setup.
- 5.** Set up each DSM-II remote port as follows:
- 5.1** At the remote site, connect laptop PC to the TERM connector on the appropriate channel bank interface card.
- 5.2** Log on to the channel bank where the remote site DSM-II card is installed. The Main Window appears.
- 5.3** Note the channel bank user slot location of the DSM-II card.
- 5.4** From the Main Window, access the DSM-II card that corresponds to intended DSM-II port.
- 5.5** Press the **Enter** key. DSM-II card is now selected. The DSM-II Card Window now appears.

**DSM-II Configuration Programming Instructions**

- 5.6** Set up the fields (along with the corresponding settings) listed in the following table for DSM-II port.

Field	Setting
STATE	stdby
SITE	rem
MODE	Set as required. (See “Mode,” page 5-24 above for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY	leave default

- 5.7** Access WAN field for the DSM-II port and assign as default setting (W1-1).
- 5.8** Access the TS field for the DSM-II port and assign it a unique time slot.
- 5.9** Set port to “actv.”
- 5.10** From the Main Window, press the **S** key to save the setup.
- 6.** Repeat steps 1 through 5.10 for any additional channels.

**Extended Broadcast Setup**

Performing an extended broadcast setup consists of four procedures:

- ☐ Master Channel Bank Setup
- ☐ Sub-Channel Bank Setup
- ☐ Broadcast Links Setup
- ☐ Remote Port Setup

Each of these procedures is described in the following sections.

**Master Channel Bank Setup**

Setting up the master channel bank creates the required prime ports and, where required, creates slave and slave-loopback (sl-lb) ports. Perform master channel bank setup as follows:

- 1.** Connect the laptop PC serial port to the TERM connector on master channel bank interface card.
- 2.** Log on to the channel bank where prime port DSM-II cards are installed. The Main Window appears.
  - 2.1** Note the channel bank user slot location of the DSM-II card(s) that is intended to provide the prime port.



**DSM-II Configuration Programming Instructions**

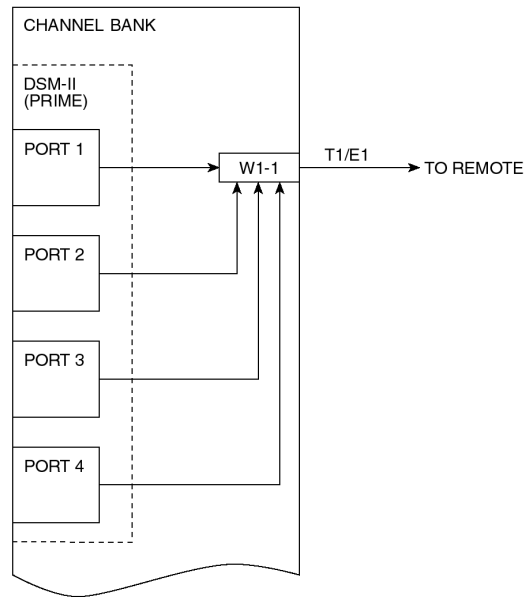
3. From the Main Window, access the DSM-II Card Window that corresponds to intended DSM-II card.
4. Set up the fields, along with the corresponding settings, listed in the following table for DSM-II card for each channel in the system.

Field	Setting
STATE	stdby
SITE	prime
MODE	Set as required. (See "Mode," page 5-24 above for more information, if required.)
M POLARITY	Set as required. (See "M-Lead Polarity," page 5-27 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default

5. The WAN and TS fields can now be set. For each DSM-II port, assign the desired WAN and time slot. The following figure shows an example of WAN assignments and the resulting port-to-WAN links. The resulting WAN T1/E1 line serves the first remote site.

## DSM-II Configuration Programming Instructions

<b>Fremont</b>	<b>: U1</b>	<b>DSM-4</b>	<b>3164</b>	<b>Rev A0-0</b>	<b>Ser 0001</b>	<b>: 01-07-99 12:49</b>
<b>Version #:1.1</b>						
	<b>1</b>	<b>2</b>	<b>3</b>	<b>4</b>		
STATE	<b>stdby</b>	stdby	stdby	stdby		
SITE	prime	prime	prime	prime		
WAN	w1-1	w1-1	w1-1	w1-1		
TS	15	16	17	18		



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6. Set ports to “actv.”
7. From the Main Window, press the **S** key to save the setup.

## DSM-II Configuration Programming Instructions

8. If the master channel bank is to contain slave port(s), set up slave port by setting DSM-II fields as listed in the following table.

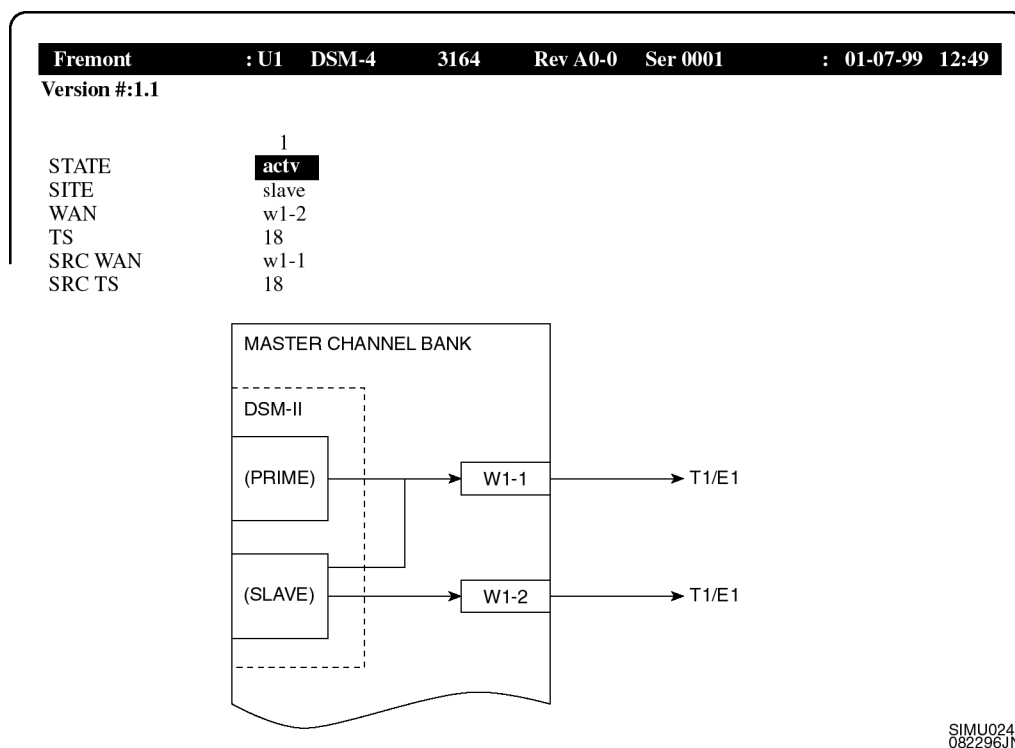
<b>NOTE</b>
-------------

Sub-channel bank usage varies from site-to-site. For systems with 14 or fewer voice channels, a master or sub-channel bank can cover two remote sites. A TeNSr 800 chassis has eight user slots. Typically, one user slot is used for an SRU card, leaving seven user slots (or 28 DSM-II ports) available. When the number of voice channels exceeds 14, a single master or sub-channel bank can only cover one remote site.

Field	Setting
STATE	stdby
SITE	slave
MODE	Set as required. (See "Mode," page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
WAN	Assign as desired
TS	Assign as desired
SRC WAN	Set to prime WAN
SRC TS	Set to prime TS
PRIME TYPE	INT

## DSM-II Configuration Programming Instructions

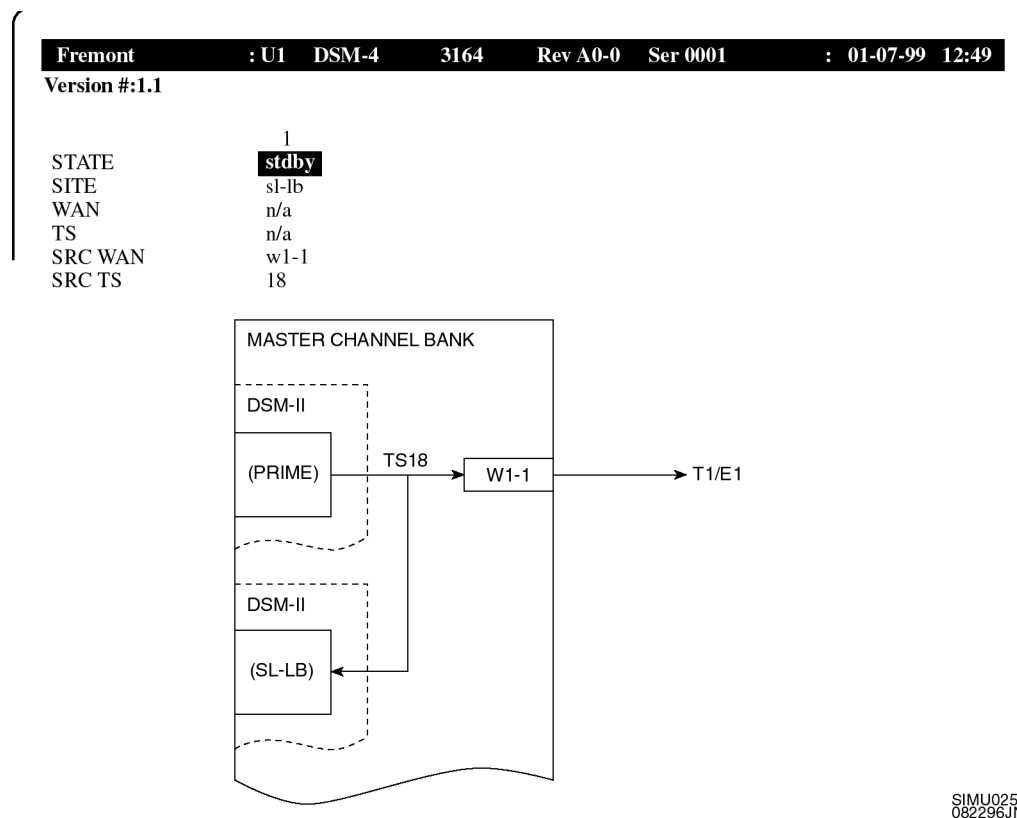
The following figure show an example of this setup, along with the resulting link.



9. Set ports to “actv.”
10. From the Main Window, press the **S** key to save the setup.
11. If the master channel bank is to cover a co-located remote site, set up the DSM-II port fields as listed in the following table.

Field	Setting
STATE	stdby
SITE	sl-lb
SRC WAN	Set same as prime WAN
SRC TS	Set same as prime TS
MODE	Set as required. (See “Mode,” page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY	leave default

The following figure shows an example of this setup, along with the resulting link.



12. On the DSM-II Card Window, set STATE field to “actv” for all ports.
13. From the Main Window, press the **S** key to save the setup.
14. Repeat steps 1 through 14 for each channel in the system.

### Sub-Channel Bank Broadcast Setup

Sub-channel bank broadcast setup creates the slave ports in the sub-channel banks. Perform sub-channel bank setup as follows:

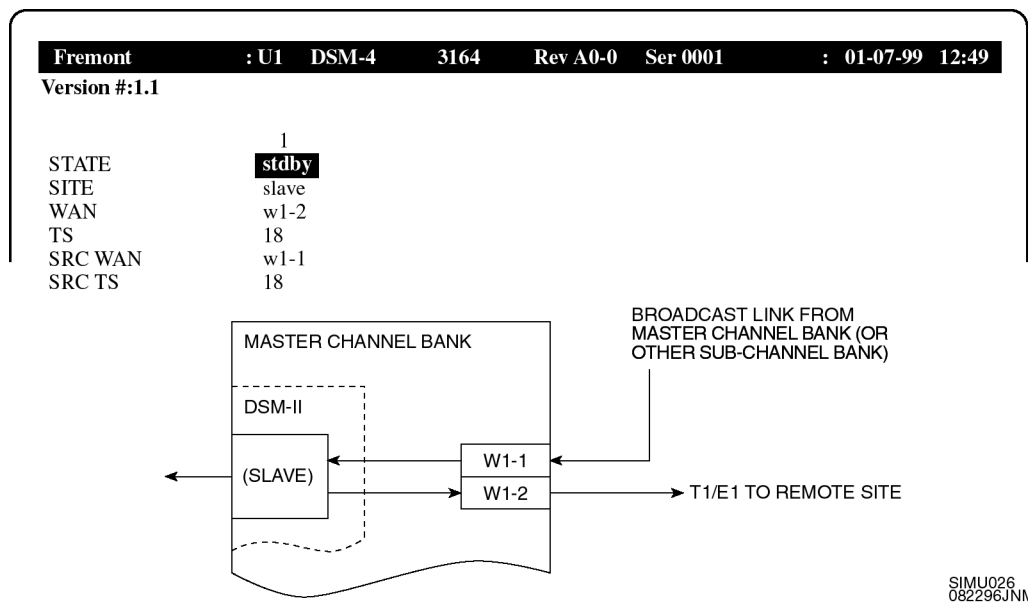
1. Connect the laptop PC serial port to the TERM connector on the appropriate sub-channel bank interface card.
2. Log on to the channel bank where DSM-II cards are installed. The Main Window appears.

## DSM-II Configuration Programming Instructions

- For non co-located remote sites, set up slave ports at the sub-channel bank by setting DSM-II port fields as listed in the following table.

Field	Setting
STATE	stdby
SITE	slave
MODE	Set as required. (See "Mode," page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
PRIME TYPE	EXT

- Set SRC WAN and SRC TS fields to the WAN and TS within the broadcast link. The broadcast link is a T1/E1 link from the master channel bank (or another sub-channel bank) that contains the broadcast of the prime port. the following figure shows an example of this SRC WAN and SRC TS assignment, along with the resulting link.



- Set ports to "actv."
- From the Main Window, press the **S** key to save the setup.

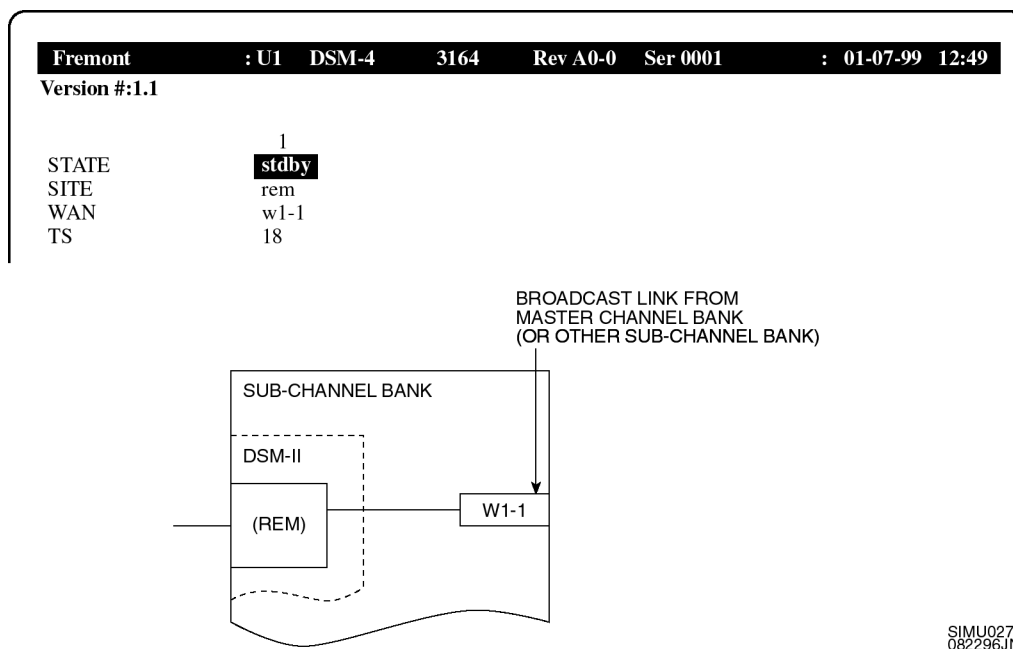
7. For co-located remote sites in a sub-channel bank, set up ports at the sub-channel bank by setting DSM-II port fields as listed in the following table.

Field	Setting
STATE	stdby
SITE	rem
MODE	Set as required. (See "Mode," page 5-24 for more information, if required.)

#### NOTE

Sub-channel bank(s) containing only DSM-II ports programmed as **rem** for use with co-located remote(s) can use channel bank equipped with CPU firmware version V3.43 or higher (hardware models 8800, 8801).

8. Set WAN and TS fields to the WAN and TS within the broadcast link. The broadcast link is a T1/E1 link from the master channel bank (or another sub-channel bank) that contains the broadcast of the prime port. The following figure shows an example of this assignment, along with the resulting link.



9. On the DSM-II Card Window, set STATE field to "actv" for all ports.
10. From the Main Window, press the **S** key to save the setup.

**DSM-II Configuration Programming Instructions**

11. Repeat steps 1 through 10 for additional channels, as required.
12. Repeat steps 1 through 11 for additional sub-channel banks, as required.

**Setting Broadcast Links**

Set broadcast links within the master channel bank as follows:

1. Where WANs will connect to sub-channel banks, set the prime port broadcast into multiple WANs as follows:
  - 1.1 Connect the laptop PC serial port to the TERM connector on the master channel bank interface card.
  - 1.2 Log on to the channel bank where DSM-II cards are installed. The Main Window appears.
  - 1.3 Access the cross-connect window of the channel bank by typing **x** from the Main Window (the window that appears after a user logs into the channel bank) of the master channel bank.

2. Press the **V** key to view all user circuits.

3. Place the cursor over the prime port to be broadcast. The prime port is displayed as a “user circuit” with user slot and port number (displayed as “U7-01”) of the DSM-II port (as shown in bold).

```
user_circuit u7-01 01*64 off w1-1 01*64 off v n/a no
```

EXAMPLE: In above display sample, port 1 of DSM-II card in user slot 7 of the channel bank is accessed.

4. Highlight the prime port user circuit to be broadcast using the up/down arrow keys.
5. With the prime port user circuit highlighted, press the **B** key (broadcast). The broadcast link dynamics (destination WAN and TS) of where the prime port will be broadcast are now shown (as shown in bold).

```
user_circuit u7-01 01*64 off w1-1 01*64 off b/v n/a no
```

Highlight the WAN field using the cursor arrow keys. (WAN field is shown in bold.)

```
user_circuit u7-01 01*64 off w1-1 01*64 off b/v n/a no
```

6. Select the WAN field where the prime port is to be broadcast, as follows:
  - 6.1 Press the **Enter** key. The WAN options are now displayed.
  - 6.2 Use the left and right arrow keys to toggle through WAN options. When the desired option is highlighted, press the **Enter** key.

7. Set the TS field where the prime port is to be broadcasted, as follows:

- 7.1 Position the cursor over the TS field (as shown in bold).

```
user_circuit u7-01 01*64 off w1-1 01*64 off b/v n/a no
```



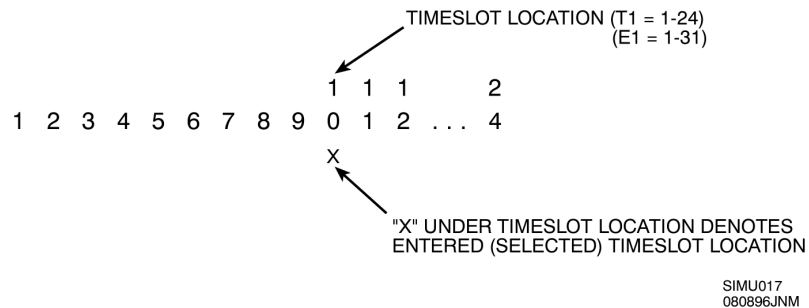
**NOTE**

Where “user\_circuit” default alias prompt appears, this alias can be renamed as desired to help identify the link (for example, “ch1 midtown”).

**NOTE**

The “01\*64” time slot field display indicates time slot size (default 01\*64 indicates 1 slot at 64k bandwidth), not selection of a specific time slot. Selection of specific time slot is done in next step.

- 7.2** Once the field is highlighted, press the **Enter** key. The time slot selection menu now appears (as shown below).

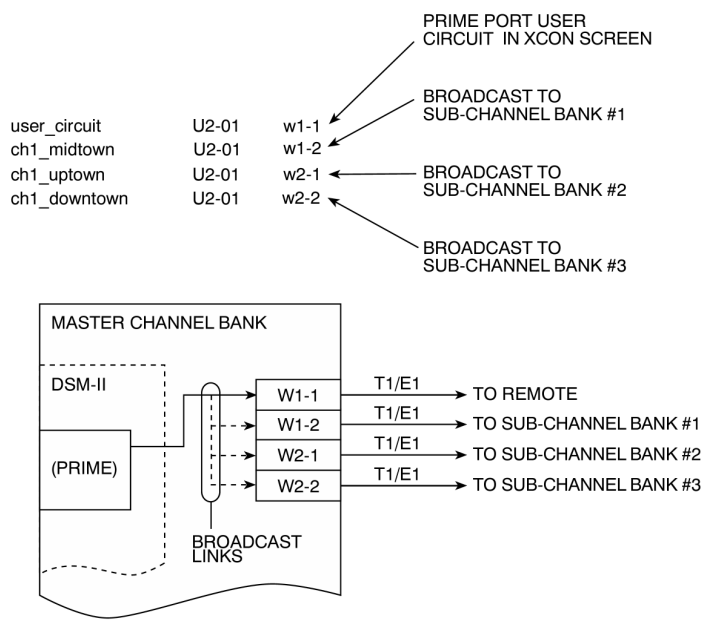


- 7.3** Using the left/right arrow keys, move the cursor to the desired time slot selection.
- 7.4** With the cursor highlighting the desired time slot, press the space bar. This selects the time slot (as denoted on the window by an “x” appearing under the selection).

**DSM-II Configuration Programming Instructions**

8. Press the **Enter** key and then press the **S** key to save the broadcast link dynamics.

The following figure shows an example of this setup, and the resulting link.

**NOTE**

Where user\_circuit default alias prompt appears, this alias can be renamed as desired to help identify the link (for example, ch1 midtown).

9. Repeat steps 3 through 8 for each master channel bank-to-subchannel bank link, as required.

**NOTE**

If subchannel bank-to-subchannel bank broadcast links are required, perform steps 10 through 15. If subchannel bank-to-subchannel bank broadcast links are not required, omit steps 10 through 15 and proceed to step 16.

10. In sub-channel bank(s) that will be broadcasted to another sub-channel bank, access the cross-connect (XCON) window of the sub-channel that will be creating the broadcast link by typing **x** from the broadcasted channel bank Main Window.

11. Press the **V** key to view all user circuits and dsmb circuits.

(A user\_circuit and its corresponding dsmb\_circuit is displayed as a pair of lines.)

```
user_circuit  u7-01  01*64  off  w1-1  01*64  off  v  n/a  no
dsmb_circuit  w1-1   01*64  off  w1-2 01*64  off  b/d n/a  n/a
```

For each user\_circuit/dsmb\_circuit window listing, the user\_circuit line identifies a slave port. The dsmb\_circuit line identifies the broadcast link that corresponds to the slave port displayed in the user\_circuit line.

12. Highlight the dsmb\_circuit of the intended slave port (user\_circuit) to be broadcast. Press the **B** key (broadcast).
13. In the dsmb\_circuit that corresponds to source slave port, enter the destination WAN and TS that creates broadcast link to another sub-channel bank as follows:

- 13.1 Position the cursor over the broadcast WAN field.

(broadcast source WAN field is shown in bold.)

```
dsmb_circuit  w1-1   01*64  off  w1-2 01*64  off  b/d n/a  n/a
```

#### NOTE

Where dsmb\_circuit default alias prompt appears, this alias can be renamed as desired to help identify the link (for example, ch1 broadcast).

- 13.2 With the WAN field highlighted, press the **Enter** key to access WAN options.

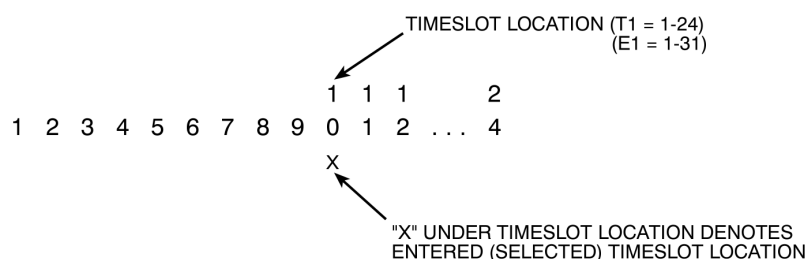
- 13.3 Position the cursor to highlight the desired WAN and press the **Enter** key.

- 13.4 In the dsmb\_circuit field, position the cursor over the TS field.

(broadcast TS field is shown in bold.)

```
dsmb_circuit  w1-1 01*64  off  w1-1  01*64  off  b/d n/a  n/a
```

- 13.5 Once the field is highlighted, press the **Enter** key. The time slot selection menu now appears.



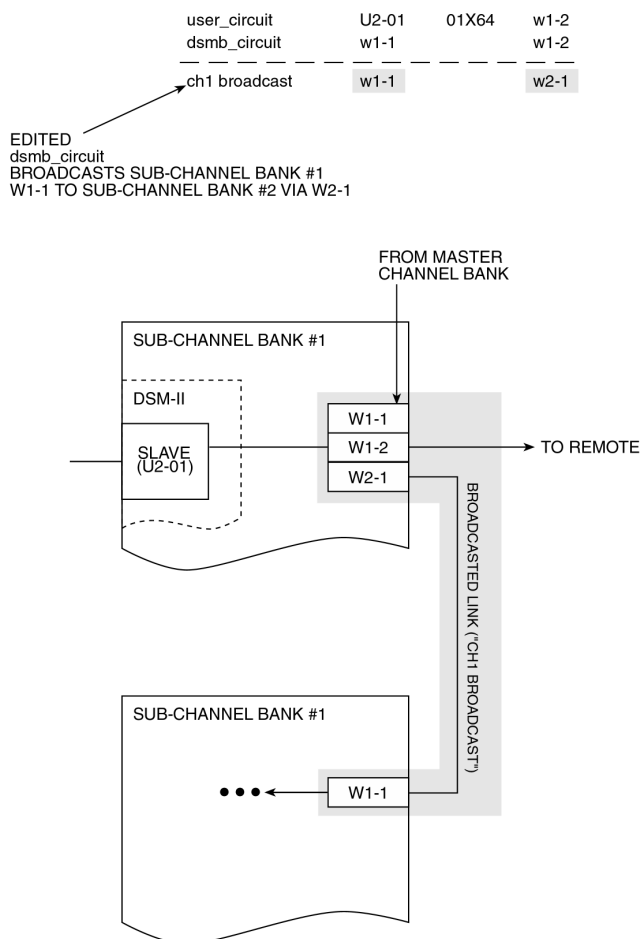
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## DSM-II Configuration Programming Instructions

**13.6** Using the arrow keys, move the cursor to the desired time slot selection.

**13.7** When the cursor highlights the desired time slot, press the space bar. This selects the time slot (as denoted on the window by an “x” appearing under the selection).

The following figure shows an example of this setup, along with the resulting link.



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- 14.** From the Main Window, press the **S** key to save the setup.
- 15.** Repeat steps 10 through 14 for any additional subchannel bank-to-subchannel bank links, as required.
- 16.** Repeat steps 1 through 15 for each channel in the system, as required.

## Remote Port Setup

Setting up the remote ports allows the remote ports to communicate with the master and sub-channel banks, as applicable. Perform remote port setup as follows:

### NOTE

Channel banks containing only DSM-II ports programmed as **rem** can use channel banks equipped with CPU firmware version V3.43 or higher (hardware models 8800, 8801).

1. Connect the laptop PC serial port to the TERM connector on the remote channel bank interface card.
2. Log on to the channel bank where remote port DSM-II cards are installed. The Main Window appears.
3. Note the channel bank user slot location of the DSM-II card that is intended to serve as remote port.
4. From the Main Window, access the DSM-II Card Window that corresponds to intended DSM-II card.
5. Set up the fields (along with the corresponding settings) listed in the following table for each DSM-II port, as applicable.

Field	Setting
STATE	stdby
SITE	rem
MODE	Set as required. (See "Mode," page 5-24 for more information, if required.)
Rx TLP CLR	leave default
Rx TLP SEC	leave default
Tx TLP CLR	leave default
MAX DELAY	leave default

6. The WAN and TS fields can now be set. For each DSM-II port, assign the desired WAN and time slot.
7. On the DSM-II Card Window, set STATE field to "actv" for all ports.
8. From the Main Window, press the **S** key to save the setup.
9. Repeat steps 1 through 8 for additional remote ports, as applicable.

---

---

## DSM-II Troubleshooting Instructions From the DSM-II Test Window

Special test modes and test signals are executed from the DSM-II Test Window. When the Test Window is accessed, the laptop PC displays several fields which, when set, execute and control the DSM-II test functions.

For each of these fields, options exist which are set to enable various test conditions and stimuli. The following sections provide an overview of the function of each Test Window field. Each section also describes how to access and set the options of each field.

### Accessing A Particular DSM-II Card From The Main Window

The Test Window works with individual DSM-II cards. Therefore, the DSM-II card to be tested must first be accessed. Access the intended DSM-II card intended for testing as follows:

1. Log on to the channel bank where the DSM-II card is installed. The Main Window appears.
2. Note the channel bank user slot location of the DSM-II card that is intended to be accessed.
3. From the channel bank Main Window, access the DSM-II card by moving the cursor to the user slot that corresponds to intended DSM-II card.
4. Press the **T** key. The Test Window is now displayed, as shown in and described in the following sections.

### Test Window Fields Overview

Figure 5-6 shows the DSM-II Test Window. Table 5-6 lists the default Test Window fields and options for the DSM-II Test Window.

The Test Window is divided into two sections:

- ☐ The first section (top half) is the user input section which starts at the **TEST** field and ends at the **E2** field.
- ☐ The second section is a status or information section which displays test values obtained and other response information. The status section starts with **FR SYNC** field and ends with the **EXS DELAY** field.

## DSM-II Troubleshooting Instructions From the DSM-II Test Window

Fremont : U4 DSM-4 3164 Rev A0-0 Ser 0001 : 01-07-99 12:49					
	1	2	3	4	
TEST	on	on	on	on	
TONE	off	off	off	off	
SEC OPT	off	off	off	off	
LB	off	off	off	off	
M1	norm	norm	norm	norm	
M2	norm	norm	norm	norm	
E1	norm	norm	norm	norm	
E2	norm	norm	norm	norm	
FR SYNC	yes	yes	yes	yes	
M1 PRES	no	no	no	no	
M2 PRES	no	no	no	no	
E1 PRES	no	no	no	no	
E2 PRES	no	no	no	no	
1PPS PRES	yes	yes	yes	yes	
MARK PRES	no	no	no	no	
RLB PRES	no	no	no	no	
DELAY	20000	20000	20000	20000	
EXS DELAY	1	1	1	1	
Save : Undo : Refresh : Main					

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Figure 5-6 DSM-II Test Window

**DSM-II Troubleshooting Instructions From the DSM-II Test Window***Table 5-6 DSM-II Test Window Fields*

Field	User Options	Default	Notes
TEST	<ul style="list-style-type: none"> <li>• off</li> <li>• on</li> <li>• n/a</li> </ul>		1
TONE	<ul style="list-style-type: none"> <li>• off</li> <li>• 1khz</li> <li>• 20hz</li> <li>• cal</li> </ul>		
SEC OPT	<ul style="list-style-type: none"> <li>• off</li> <li>• sqr</li> <li>• stair</li> <li>• randm</li> </ul>		
LB	<ul style="list-style-type: none"> <li>• off</li> <li>• lvf</li> <li>• rlb</li> </ul>		
M1	<ul style="list-style-type: none"> <li>• norm</li> <li>• on</li> <li>• off</li> </ul>		
M2	<ul style="list-style-type: none"> <li>• norm</li> <li>• on</li> <li>• off</li> </ul>		
E1	<ul style="list-style-type: none"> <li>• norm</li> <li>• on</li> <li>• off</li> </ul>		
E2	<ul style="list-style-type: none"> <li>• norm</li> <li>• on</li> <li>• off</li> </ul>		
FR SYNC	—	—	2
M1 PRES	—	—	2
M2 PRES	—	—	2
E1 PRES	—	—	2
E2 PRES	—	—	2
1 PPS PRES	—	—	2
MARK PRES	—	—	2
RLB PRES	—	—	2
DELAY	—	—	2
EXS DELAY	—	—	2
<b>NOTE:</b> 1. N/A is displayed when a port is in standby. 2. These fields are displays only with no user entry. ***** is displayed when a port is in standby. N/A is displayed when test mode is turned off.			



## Test Window Fields Detailed Information

Each of the Test Window fields and their uses are described in the following sections. Instructions on how to access and set each field are also provided.

### Test

The Test field is used to selectively enable or disable testing for the ports on the entered DSM-II card. Set this field as follows:

1. Select the desired port(s) by placing cursor over the TEST field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select **on** or **off** for each port. When the desired setting is displayed, press the **Enter** key.
3. Press the **S** key to save changes.

#### NOTE

If the port is in standby, N/A is displayed for the test field.

### Tone

The Tone field generates test tones. The options for the Tone field are:

- ☐ 1khz
- ☐ 20hz
- ☐ cal

The **1khz** option generates a 1 kHz, -10 dBm tone. The **20hz** option generates a 20 Hz, -10 dBm tone. The **cal** option generates a 1 kHz, 0 dBm tone. When a test tone is turned on, it can monitored on the receive side of the port only (test tones are local only).

#### NOTE

Test tone dBm values are relative to 0 dB “Rx TLP CLR” setting.

The 1 kHz tone is used for coarse level adjustment. The 20 Hz tone is used for the fine amplitude adjustment.

The output level for the tone options are adjusted through the “Rx TLP CLR” only since the tones are generated out the receive side of the port. The tones are used for setting line levels, coarse deviation, and fine tune amplitude adjustments.

**DSM-II Troubleshooting Instructions From the DSM-II Test Window**

Activate test tones as follows:

1. From the Test Window, move the cursor with the arrow keys to highlight the TONE field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired tone. When the desired setting is displayed, press the **Enter** key.
3. Press the **S** key to save changes.

**SEC OPT (SECURENET Optimization)**

The Sec Opt field generates SECURENET optimization patterns. The options for the sec opt field are:

- ☐ **sqr** (square wave)
- ☐ **stair**
- ☐ **randm** (random)

The **sqr** pattern generates a 1.5 kHz square wave. This pattern is used for setting the SECURENET deviation.

The **stair** pattern generates a staircase pattern. This pattern is used for checking for polarity inversions on the 600 $\Omega$  lines.

The **randm** pattern generates a random data pattern at the 6 kbaud rate for 4L systems and 12 kbit for 2L systems. This pattern is used for manually checking the SECURENET phasing.

**NOTE**

All sec opt patterns is activated on the prime port only.

Activate SEC OPT pattern as follows:

1. On the prime port of the desired channel, highlight the SEC OPT field. Press the **Enter** key.
2. Use the arrow keys to highlight the TONE field. Press the **Enter** key.
3. Use the up and down arrow keys to select the desired SEC OPT pattern. When the desired setting is displayed, press the **Enter** key.
4. Press the **S** key to save changes.

## LB (Loopback)

The lb field is used to place a given port in loopback for testing purposes. The options for this field are:

- ☐ lvf (local voice frequency)
- ☐ rlb (remote loopback)

When a port is placed in local voice frequency loopback, the loopback will occur at the local DSM-II port. Audio that is injected into the transmit side of the port is looped internally to the receive side of the port.

When a port is placed in remote loopback, the loopback will occur at the far end DSM-II port. Audio that is injected into the transmit side of the port is routed down the T1 or E1 line to the far end DSM-II port a looped back down the T1 or E1 line to the receive side of the DSM-II port.

### NOTE

A **slave** and **sl-lb** port must be temporarily reprogrammed as a **rem** port when using the loopback test modes.

When a port is placed in remote loopback, the corresponding far end port will indicate remote loopback present. This is indicated in the Test Window of the far end port.

### IMPORTANT:

Software release V2.9 (or higher) will NOT allow remote loopback with the test mode turned off in the far-end port. The Test Field mode must be turned on in both ports.

Activate loopback mode as follows:

1. From the Test Window, use the arrow keys to highlight the LB field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired loopback option for the selected port. When the desired setting is selected, press the **Enter** key.
3. Press the **S** key to save changes.

---

DSM-II Troubleshooting Instructions From the DSM-II Test Window**M1**

The M1 field is used to provide manual control of the M1 Lead. The options for this field are:

- ☐ norm
- ☐ on
- ☐ off

When the M1 field is set to **norm**, the M1 Lead will operate normally. With the normal option selected, activation of the M1 Lead only occurs via the M1 Lead interface (grounding of the M1 input).

When the M1 field is set to **on**, the M1 Lead is activated independent of the M1 Lead's input. When the M1 field is set to **off**, the M1 Lead cannot be activated through the M1 Lead interface.

Activate or de-activate the M1 Lead as follows:

1. From the Test Window, use the arrow keys to highlight the M1 field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired option for the M1-Lead. When the desired setting is selected, press the **Enter** key.
3. Press the **S** key to save changes.

**M2**

The M2 field is used to provide manual control of the M2 Lead. The options for this field are:

- ☐ norm
- ☐ on
- ☐ off

When the M2 field is set to **norm**, the M2 Lead operates normally. With the normal option selected, activation of the M2 Lead only occurs via the M2 Lead interface (grounding of the M2 input).

When the M2 field is set to **on**, the M2 Lead is activated independent of the M2 Lead's input.

When the M2 field is set to **off**, the M2 Lead cannot be activated through the M2 Lead interface.

<b>NOTE</b>
-------------

If the port is programmed for the 2L or 4L SECURENET modes, then activation of the M2 Lead either through the M2 Lead interface or the Test Window will place the port in the SECURENET mode of operation.

---

Activate or de-activate the M2 Lead as follows:

1. From the Test Window, use the arrow keys to highlight the M2 field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired option for the M2-Lead. When the desired setting is selected, press the **Enter** key.
3. Press the **S** key to save changes.

## E1

The E1 field is used to provide manual control of the E1 Lead. The options for this field are:

- ☐ norm
- ☐ on
- ☐ off

When the E1 field is set to **norm**, the E1 Lead operates normally. With the Normal option selected, activation of the E1 Lead occurs only via activation of the M1 Lead.

When the E1 field is set to **on**, the E1 Lead is activated independent of the M1 Lead's input.

When the E1 field is set to **off**, the E1 Lead will not activate when the M1 Lead goes active.

<b>NOTE</b>
-------------

The Test Window will show the E1-Lead being active, although the relay will not activate.

Activate or de-activate the E1 Lead as follows:

1. From the Test Window, use the arrow keys to highlight the E1 field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired option for E1 Lead. When the desired setting is selected, press the **Enter** key.
3. Press the **S** key to save changes.

**DSM-II Troubleshooting Instructions From the DSM-II Test Window****E2**

The E2 field is used to provide manual control of the E2 Lead. The options for this field are:

- ☐ norm
- ☐ on
- ☐ off

When the E2 field is set to **norm**, the E2 Lead will operate as normal. With the Normal option selected, activation of the E2 Lead occurs only via activation of the M2 Lead.

When the E2 field is set to **on**, the E2 Lead is activated independent of the M2 Lead's input.

When the E2 field is set to **off**, the E2 Lead will not activate when the M2 Lead goes active.

**NOTE**

Activating the E2 Lead will not place a port that is programmed for 2L or 4L mode into the SECURENET mode.

Activate or de-activate E2 Lead as follows:

1. From the Test Window, use the arrow keys to highlight the E2 field of the desired port. Press the **Enter** key.
2. Use the arrow keys to select the desired option for the E2 Lead. When the desired setting is selected, press the **Enter** key.
3. Press the **S** key to save changes.

**FR Sync**

The Frame Synchronization field is an information field only; there is no user input. This field indicates that the port under test is receiving the DSM-II frame structure from the far-end DSM-II port. The possible messages are:

- ☐ n/a (port is not in test mode)
- ☐ yes (port has frame sync)
- ☐ no-w (port does not have frame sync)

**NOTE**

This field does not indicate that the DSM-II port at the far end is receiving frame structure properly.

### M1 Pres

The M1 Presence field is an information field only; there is no user input. This field indicates the state of the M1 Lead. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (M1 Lead not active)
- ☐ yes (M1 Lead is active)

### M2 Pres

The M2 Presence field is an information field only; there is no user input. This field indicates the state of the M2 Lead. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (M2 Lead not active)
- ☐ yes (M2 Lead is active)

### E1 Pres

The E1 Presence field is an information field only; there is no user input. This field indicates the state of the E1 Lead. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (E1 Lead not active)
- ☐ yes (E1 Lead is active)

### E2 Pres

The E2 Presence field is an information field only; there is no user input. This field indicates the state of the E2 Lead. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (E2 Lead not active)
- ☐ yes (E2 Lead is active)

---

DSM-II Troubleshooting Instructions From the DSM-II Test Window

### 1 PPS Pres

The 1 PPS Presence field is an information field only, there is no user input. This field indicates that the DSM-II is receiving a local 1 PPS signal from the GPS receiver. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (no local 1 PPS present)
- ☐ yes (local 1 PPS present)

<b>NOTE</b>
-------------

The **yes** response does not guarantee that the GPS receiver is locked to the GPS satellite system. GPS lock indication is provided by GPS receiver.

---

### Mark Pres

The Mark Presence field is an information field only; there is no user input. When the prime port receives its local 1 PPS signal, it inserts a marker in the DSM-II frame structure that is transmitted out to the remote site DSM-II port. This field indicates whether the marker from the prime port is being decoded at the remote site DSM-II port. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (no marker present)
- ☐ yes (marker present)

<b>NOTE</b>
-------------

This field applies only to **rem**, **lb**, and **sl-lb** ports only.

---

### RLB Pres

The RLB Presence field is an information field only; there is no user input. This field indicates that the DSM-II port has been placed in remote loopback by the far-end DSM-II port. The possible messages are:

- ☐ n/a (test mode is inactive)
- ☐ no (no remote loopback present)
- ☐ yes (remote loopback present)



## Delay

The delay field is an information field only; there is no user input. This field indicates whether or not the phase delay is being calculated properly. When the phase delay is being calculated properly, the "DELAY" reading should be within 1.0 to 3.0  $\mu$ sec of the "MAX Delay" setting. The possible messages are "n/a" (test mode is inactive) or the integer value of the delay (in  $\mu$ sec). (The integer value will only appear when the test mode is on, the port is receiving its local 1 PPS from a GPS receiver, and the port is receiving the marker from the prime port.)

### NOTE

This field applies only to **rem**, **lb**, and **sl-lb** ports only.

## EXS Delay

The EXS (Excess) Delay field is an information field only; there is no user input. For a given MAX Delay setting, the excess delay field shows the amount of extra delay in the path. The excess delay is the additional delay added to the path so that the MAX Delay is achieved. Using this data, the MAX Delay setting is tailored accordingly. The excess delay is displayed in msec and has a  $\pm 1.0$  msec accuracy. If the EXS Delay field is reporting an alarm, then the path delay is greater than the MAX Delay. When a port is in alarm, the audio output is muted and the E-Leads are released. In order to bring the port out of alarm, the MAX Delay must be increased until the Excess Delay field displays an integer value.

### NOTE

This field applies only to **rem**, **lb**, and **sl-lb** ports only.

When the DSM-II port is operating in the SECURENET mode (port programmed for SECURENET and the M2 Lead is active), the Excess Delay is approximately 1 to 2 msec larger.

**DSM-II Troubleshooting Instructions From the DSM-II Test Window****DSM-II Test Window Editing Options**

At the bottom of the DSM-II Test Window, there are several generic editing options that are used when running the test mode. Table 5-7 lists these editing options.

Table 5-7 **DSM-II Test Window Options**

Option	Function
Save	Saves changes to settings. When changes are made in the Test Window, an asterisk appears next to the port number where the changes were made. This asterisk indicates that changes were made but not saved. Press the <b>S</b> key to save changes.
Undo	Returns all settings to the last saved state. Press the <b>U</b> key to return to last saved state.
Refresh	The Test Window is not real time. By refreshing the Test Window, the current status of the Test Window fields is displayed. Press the <b>R</b> key to refresh the Test Window.
Main	Returns to the DSM-II Main Window of the channel bank. Press the <b>M</b> key to return to the Main Window of the channel bank.

## DSM-II Phasing Using 1 PPS (GPS Operation)

The following section describe how the DSM-II cards at the local and remote ends of a network channel maintain proper phasing.

### DSM-II Phasing Operation

GPS operation (1 PPS phasing) commences when there is a local 1 PPS source at the prime site and the remote site. With synchronized 1 PPS signals at the local and remote ends, the throughput delay can be quantized and controlled. The total throughput delay from the 600 $\Omega$  input at the prime site to the 600 $\Omega$  output at the remote site is equal to the **MAX Delay** field setting. The throughput delay value is a sum of three parts, as follows:

- ❑ existing network delay
- ❑ DSM-II card delay
- ❑ DSM-II buffer delay, which is proportionally variable to any changes in network delay. The buffer delay tracks the network delay changes and adjusts itself accordingly to maintain a constant delay value (MAX Delay).

The components of the throughput delay are shown in Figure 5-7.

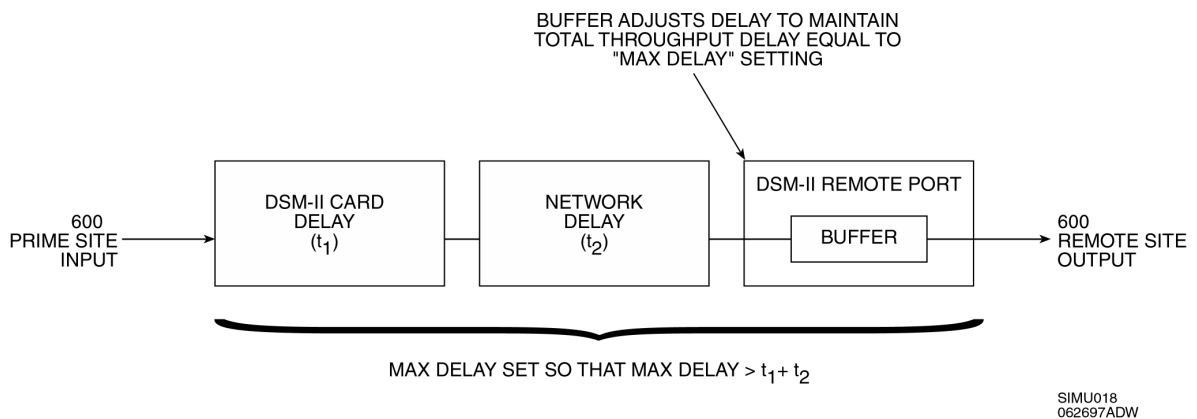


Figure 5-7 **Total Throughput Delay And Buffer Delay Adjustment**

The prime site applies a time stamp marker to the outgoing data, which is received by the remote-site DSM-II card. With both the prime-site signal arrival time and the local 1 PPS source arrival time being known, the DSM-II calculates the path delay and adjusts its buffer so that the delay from the 600 $\Omega$  input to the 600 $\Omega$  output is equal to the MAX Delay value. This measurement and adjustment is repeated at 1 second intervals, with the 600 $\Omega$  input-to-600 $\Omega$  output delay being adjusted to yield the programmed MAX Delay value.

**DSM-II Phasing Using 1 PPS (GPS Operation)****NOTE**

The remote site DSM-II port must have a local 1 PPS present and the marker present (time stamp from prime DSM-II) in order to adjust the buffer. If either the 1 PPS or the marker is not present, then the DSM-II will not change the buffer size. In this condition, the buffer will maintain its setting from the last valid time stamp interval.

**GPS Input (1 PPS Input) Impedance Setting**

The input impedance for the DSM-II 1 PPS connector is set via a jumper for either  $50\Omega$  or high impedance ( $1000\Omega$ ). The jumper is located next to the BNC connector for the 1 PPS input and is labeled JP2. With the jumper installed, the input impedance is  $50\Omega$ . With the jumper out, the input impedance is  $1000\Omega$  (HI-Z). (The jumper is normally installed.) When a 1 PPS signal is connected to a DSM-II card in a given channel bank, all cards within the channel bank are connected to the 1 PPS signal.

When there are multiple channel bank enclosures that contain DSM-II cards and a limited number of 1 PPS sources are available, then a single 1 PPS source can be daisy-chained between several channel banks. In this case, a single DSM-II card is used to provide a  $50\Omega$  termination (jumper in) for the GPS receiver while the remaining DSM-II cards are set for high impedance (jumper out). BNC T-connectors are used to daisy-chain the single 1 PPS signal between the GPS receiver and the DSM-II cards. Figure 5-8 shows an example of the daisy-chain connection.

**NOTE**

No more than five DSM-II cards (including DSM-II card set for  $50\Omega$ ) should be driven from a single 1 PPS signal source.

On the Rev A DSM-II hardware, the 1 PPS input impedance is  $50\Omega$  only. All other DSM-II hardware version will have the 1 PPS input impedance jumper. The hardware version of the DSM-II is determined by looking at the label (label reads "REV XX") on the back of the DSM-II board or by looking at the top of the DSM-II Main Window (displayed as "REV XX-0").

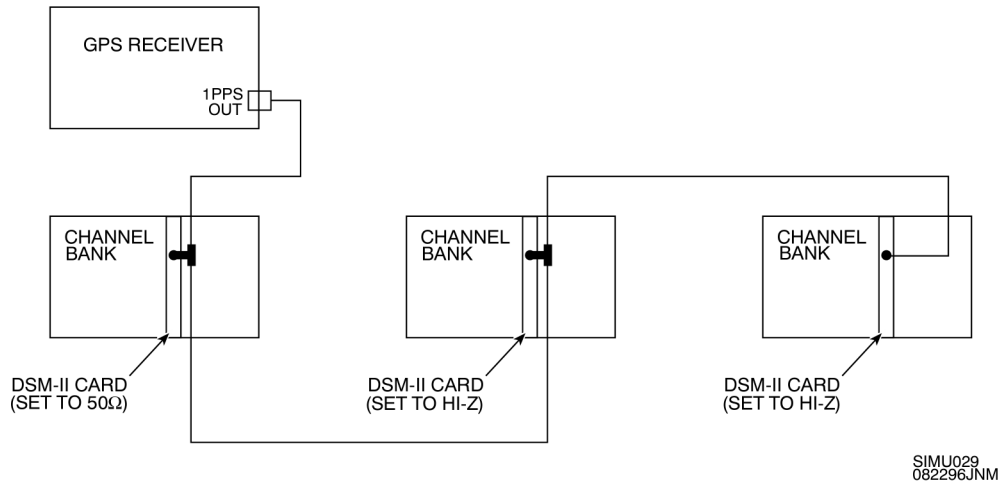
**DSM-II Phasing Using 1 PPS (GPS Operation)**

Figure 5-8 **1 PPS Connections For Multiple DSM-II Cards**

## **DIGITAC Channel Immediate Disable for T1/E1 Failures (Optional)**

In normal operation, the DIGITAC comparator will unsquelch and vote a channel when the status tone is removed. If the level of the input signal is less than the status tone, the DIGITAC will fail the channel. The time that the DIGITAC waits to fail the channel is specified by the Line Failure Time. This parameter has a default value of 60 msec and can be set as low as 15 msec.

In a similar manner, if a T1/E1 line goes down, the DIGITAC will unsquelch and erroneously vote the channel due to loss of status tone. The DSM-II is designed to mute the audio output and release the signaling leads (E-Leads) when the T1/E1 line goes down or the T1/E1 reaches a Bit Error Rate (BER) of 1E-3. However, when the DSM-II mutes its output, the noise characteristics are such that the DIGITAC will unsquelch and vote that channel over valid channels until the line failure time has expired. This false voting condition can interfere with the operation of the entire system. In order to avoid the false voting interval, DIGITAC inputs can be set up to disable immediately upon DSM-II muting due to a problem with the T1/E1 line.

The immediate disable feature is implemented by grounding the M1-Lead signaling lead at the remote site (see Figure 5-9). Grounding the M1-Lead signaling lead at the remote site allows for grounding of the DIGITAC fail/disable line during T1/E1 problems as described in the next section.

At the prime site, the E1 normally closed signaling lead is connected to the DIGITAC fail/disable line. For each signaling lead (E1 and E2), the DSM-II has two types: a normally closed and a normally open relay. Normally closed and normally open are the respective states of the E-Lead relays when the M-Lead is not active.

**DSM-II Phasing Using 1 PPS (GPS Operation)**

With the M1-Lead grounded, the E1 normally closed relay will be open, thereby allowing for normal operation of that DIGITAC receiver input. However, as soon as the T1/E1 goes down or the T1/E1 BER reaches  $1E-3$ , the DSM-II will release the E-Lead relay which, in turn, will ground the fail/disable line of the DIGITAC via the normally closed E1 relay.

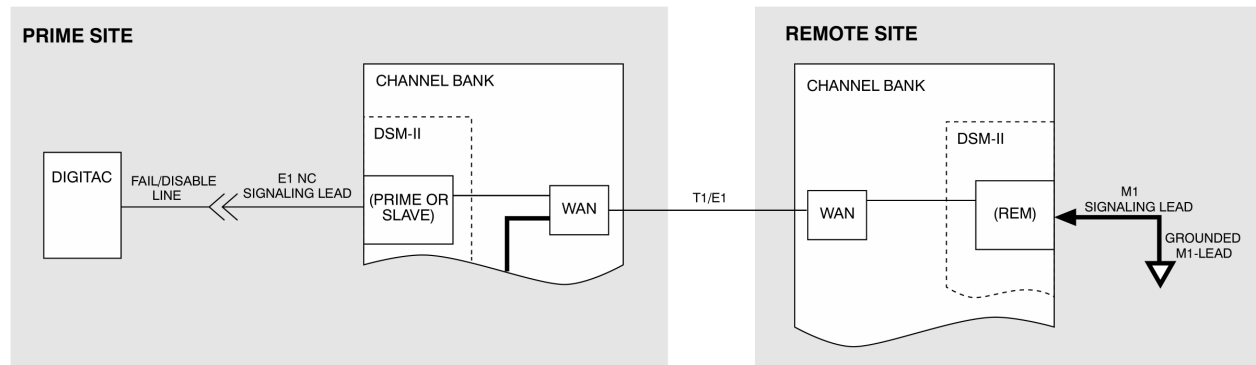
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Figure 5-9 **Immediate Channel Disable Setup**

## DSM-II Pinouts

Figure 5-10 shows the pin arrangement on the DSM-II telco connector. Table 5-8 lists the connector pinout and signal assignments for the DSM-II telco connector.

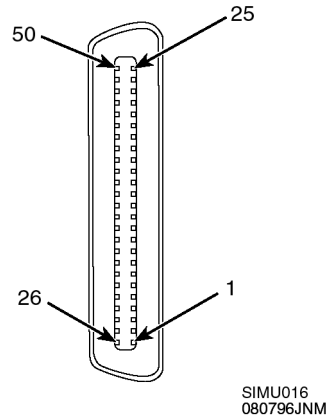


Figure 5-10 **DSM-II Telco Connector Pin Arrangement**

Table 5-8 **DSM-II Card Pin Assignment**

Port No.	Pin No.	Signal
1	1	R
	26	T
	2	R1
	27	T1
	3	M1
	28	M2
	4	ENO1
	29	ENC1
	5	EC1
	30	EC2
	6	ENO2
	31	ENC2

**DSM-II Pinouts***Table 5-8 DSM-II Card Pin Assignment (continued)*

Port No.	Pin No.	Signal
2	7	R
	32	T
	8	R1
	33	T1
	9	M1
	34	M2
	10	ENO1
	35	ENC1
	11	EC1
	36	EC2
	12	ENO2
	37	ENC2
3	13	R
	38	T
	14	R1
	39	T1
	15	M1
	40	M2
	16	ENO1
	41	ENC1
	17	EC1
	42	EC2
	18	ENO2
	43	ENC2



Table 5-8 **DSM-II Card Pin Assignment (continued)**

Port No.	Pin No.	Signal
4	19	R
	44	T
	20	R1
	45	T1
	21	M1
	46	M2
	22	ENO1
	47	ENC1
	23	EC1
	48	EC2
	24	ENO2
	49	ENC2
All Channels	25	GND
	50	GND

**NOTE:** Abbreviations in table are:

R – Transmit Ring (Mod in Ring)  
T – Transmit Tip (Mod in Tip)  
R1 – Receive Ring (Demod out Ring)  
T1 – Receive Tip (Demod out Tip)  
M1 – M1 Lead  
M2 – M2 Lead  
ENO1 – E Lead Normally Open 1  
ENC1 – E Lead Normally Closed 1  
ENO2 – E Lead Normally Open 2  
ENC2 – E Lead Normally Closed 2  
EC1 – E Lead Common 1  
EC2 – E Lead Common 2

**DSM-II Pinouts**

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# ***Data Broadcast Box***

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## **Overview**

This chapter provides a description of the DB-25 Data Broadcast Box (DBB). The following topics are included in this chapter:

<b>Topic</b>	<b>Page</b>	<b>Description</b>
Description	6-2	Describes the DBB role in the system. Describes DBB operation, indicators, and specifications.
Installation	6-4	Describes how to install the DBB.
Troubleshooting	6-5	Provides DBB troubleshooting information and corrective action.

**Description**

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**Description**

The DB-25 Data Broadcast Box (DBB) is a line-sharing device that can receive an asynchronous V.24 (EIA-232) signal and redistribute it to up to eight slave devices. One or more of the slave devices can be another DBB. The DBB also acts as a data concentrator, receiving a signal from one of the eight devices and routing it to a single V.24 output.

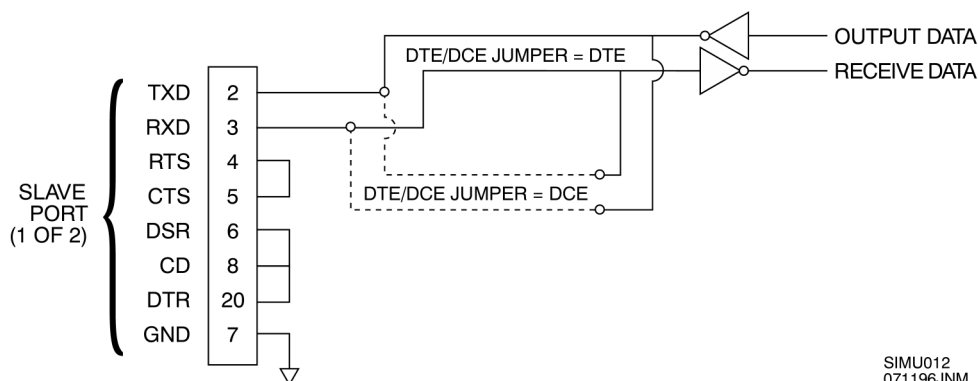
The DBB is used in several places throughout the SmartZone<sup>®</sup> system and is used as follows:

- ❑ Concentrates diagnostics and distributes the control signals from multiple DIUs into a single Zone Controller (ZC) port.
- ❑ Concentrates diagnostics and distributes the control signals between a 6809 controller and QUANTAR<sup>®</sup>/QUANTRO<sup>®</sup> stations.
- ❑ At the prime site of a voting or simulcast system, concentrates and distributes control data to the voting comparator for each channel.

**Signal Processing**

The DBB sends and receives data at each port connector over transmit data (TXD) and receive data (RXD) lines. The input to the composite port (labeled COMP on the front panel) is regenerated and simultaneously sent out on the output lead of each of the eight slave ports, channels 1 through 8.

The inputs into each of the slave ports are regenerated, OR'd together, and assembled to produce the output signal, which is transmitted from the COMP port. The signal grounds for all of the ports are tied together. RTS, CTS, DTR, DSR, and CD are all supported and are looped back. (see Figure 6-1).



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Figure 6-1 DBB Pinouts And Signal Routing

## Front Panel Indicators

The DBB is equipped with LED front panel indicators as described in Table 6-1.

Table 6-1 **Front Panel Indicator Functions**

Front Panel Indicator	Function
Power (PWR) LED	Illuminates when primary power is applied to DBB.
Composite (COMP) LED	Illuminates whenever a "space" is detected on composite input signal data path. In normal use, COMP LED will flicker, verifying data stream consisting of marks and spaces.
CH1-8 LEDs	Illuminate whenever a "space" is detected on slave port input. In normal use, CH1-8 LEDs will flicker, verifying data stream consisting of marks and spaces.

## Specifications

The DBB specifications are listed in Table 6-2.

Table 6-2 **Data Broadcast Box Specifications**

Item	Specification
Power Input: Model TL158 Model TL158-E	115 VAC, 60 Hz at 50 mA 230 VAC, 50 Hz at 25 mA
Power Consumption:	6 W
Size:	2.5" H X 12.25" W X 11.125" D (6.35 cm X 31.1 cm X 28.26 cm)
Weight:	7 lb., 2.3 oz. w/o power supply (3.24 kg) 8 lb., 5.1 oz. w/ power supply (3.77 kg)
Connectors:	9 DB-25S (female)
Interface:	Each port individually DTE/DCE selectable.
Transmission Rate:	Up to 19.2 kbps. (The DBB is transparent to data rate.)

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**Installation**

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**Installation**

Install the DBB as follows:

1. Rack mount the DBB using optional rack mount kit, PN TL485.
2. Connect the data source to the COMP (composite) connector on the rear of the DBB. Depending on system configuration and DBB usage, connect DBB to one of the following data sources:
  - The port on the Zone Controller configured for DIU control
  - The 6809 CSC\_TSC port and ACB\_SER5 connectors using “Y” cable, PN TKN9259A or TKN9260A
  - The upstream DBB
3. Set the DBB DTE/DCE jumpers to the DTE setting.
4. Connect up to eight slave devices to the DBB.
5. Plug the wall-mount transformer into 115 VAC receptacle. The red PWR (power) light on the DBB illuminates.
6. Apply power to the master and slave devices. The DBB is now ready for operation.

## Troubleshooting

The most effective way to troubleshoot a suspected problem with the DBB is to first examine the state of the LEDs on the front panel of the unit. The LEDs illuminate when each slave device connected to the DBB is transmitting data.

### Normal indications

During normal operation of a lightly loaded system, the LEDs on the DBB front panel should illuminate in the following sequence:

The COMP LED blinks, followed by a blink of port LED 1.

The COMP LED blinks, followed by a blink of port LED 2.

This continues in succession up to port LED 8.

Blinks of the COMP LED continue to alternate with blinks of the port LEDs corresponding to the slave devices connected to the DBB. The polling device (ZC or prime site 6809) continues this pattern while it sequentially polls all devices connected to the DBB.

When user voice channel traffic is present, the steady polling will appear to become slightly more random.

### Problem indications

Table 6-3 describes DBB error indications related to installation problems and the corresponding corrective action for each error.

Table 6-3 **DBB Installation Troubleshooting Procedure**

Error Indication	Corrective Action
The COMP LED or any port LED is constantly illuminated.	Affected port has device that is continuously transmitting on the bus and blocking other devices from using the shared link. This results in temporary failure of all devices connected to this particular DBB.
LEDs corresponding to ports with devices connected never blink.	Device is not responding to polls. Check for the following and repair as required: <ul style="list-style-type: none"> <li>• Broken Tx data line</li> <li>• Broken Rx data line</li> <li>• Carrier detect false</li> <li>• Improper polled device address/channel number assigned</li> <li>• Defective DIU, station, and/or total area coverage defective</li> </ul>
Two or more port LEDs blink simultaneously.	More than one device is programmed to the same polled address, resulting in controller (ZC or 6809) reporting both of the remote devices as failed. Correct by ensuring that the polling address of each of the remote devices is unique.

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**Troubleshooting****Out-of-Service Testing**

Comprehensive out-of-service testing can be done using a dumb terminal or a protocol analyzer. In accordance with dumb terminal or protocol analyzer instructions, insert typed characters or protocol analyzer test patterns on one side of the DBB and verify that they are correctly received on the other side.





# Optimization

## Overview

This chapter describes optimization procedures for all three types of Simulcast systems: analog, digital, and mixed. This chapter consists of the following topics:

Section	Page	Description
Why Simulcast Optimization is Required	7-2	Describes why optimization is required. Discusses link optimization techniques and systems.
Optimization Requirements	7-8	Describes when and how to optimize digital or analog systems.
Analog System Optimization	7-9	Provides procedures for optimizing analog channels.
Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)	7-57	Provides procedures for optimizing inbound links on digital (ASTRO <sup>®</sup> ) channels using delay measurement and optimization procedures. Individual procedures are provided for various link types.
Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping)	7-71	Provides optimization procedure for systems utilizing ASTRO-TAC 3000 with Ping comparator.
Digital (ASTRO) System Outbound Optimization	7-75	Provides procedures for optimizing outbound (station to mobile) simulcast operation in ASTRO systems using the ASTRO-TAC <sup>®</sup> /ASTRO-TAC 3000 comparator.
Mixed System Optimization	7-82	Provides references to procedures where mixed systems are used.
Performance Verification	7-82	Describes methods of system performance verification.
Optimization Using A Remote Console	7-83	Provides information on using the remote console.

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## Why Simulcast Optimization is Required

In order for the simulcast system to achieve complete coverage of its intended geographic operating area, certain multiple-coverage areas will exist where a receiving mobile or portable radio can detect transmissions from more than one site. In addition, in portions of multiple coverage areas, the relative signal strengths of multiple received transmissions should be close enough so that the mobile is not captured by any one of the station transmitters.

Since the information carried over incoming simulcasted signals is identical, the recovered signals may interfere with each other. Distortion results unless the recovered signals are in-phase and of equal amplitude. Ideally, the signals should be exactly in-phase; however, in practice, some variation is tolerable without significant distortion. For analog systems, amplitude adjustment is also required to avoid distortion in the non-capture area.

### DSM-II Analog Systems Phasing

The DSM-II module performs real-time measurement of the system transport time delay per individual DSM-II port by the use of GPS time transfer techniques. The DSM-II module receives the 1 PPS GPS signal from a local GPS receiver. With synchronized 1 PPS signals at the local and remote ends, the throughput delay can be quantified and controlled.

The DSM-II module can automatically maintain a preprogrammed time delay, independent of the actual communication path taken from the simulcast prime site to the simulcast remote site. This allows the DSM-II to be routed through Digital Crossconnect Switches (DCS) and through paths that have inconsistent delays (such as alternate path routing schemes).

The DSM-II buffer delay is proportional to the network delay. The DSM-II contains a delay buffer that tracks the network delay changes and adjusts itself accordingly to maintain a constant delay value.

### ASTRO Digital Systems Phasing

The ASTRO simulcast base stations work in conjunction with the ASTRO-TAC/ASTRO-TAC 3000 comparator to automatically transmit the digital information from all remote site stations at exactly the same time. The comparator receives a 1 PPS GPS signal from a local GPS receiver. An identical 1 PPS signal is also received by a GPS receiver located at the remote transmitter site. With the same time reference at each end of the link and a time to launch offset from the 1 PPS sent along with the digital audio from the comparator, the exact time to launch the signal is known by the station.

## Inbound Link Delay Optimization (ASTRO Only)

### Reason for Inbound Optimization

Signals arriving at geographically separated receivers are received at different times due to the differing distances that the signals must travel. The total link delay is affected by the time delay incurred over the RF channel from subscriber to satellite receiver, and by the time delay caused by the infrastructure transmission path from satellite receiver to comparator. The processing delay of the receivers affects the infrastructure delay. Infrastructure components, such as modems or channel banks, can also affect the delay incurred by a signal traveling from a subscriber to a comparator. The delay from all receivers sending signals to the comparator must be equalized (optimized) in order for the comparator to correctly vote.

### Differential Considerations in Link Delays

More important than a particular delay figure for any given path is the relative delay figure obtained when comparing the delay on one path versus another. This relative delay figure between paths is referred to as *differential* delay. This difference in the path delay must be determined. In order for the comparator to properly vote, these delays must be evaluated to make certain they fall within certain limits. The differential limits and the delay components are discussed in the following section.

### Link Delay Components and Comparator Limits

The total delay a link experiences consists of both a *fixed, known* component and an *unknown* component, which may be either fixed or variable.

The fixed component of a delay is compensated for by the comparator using Radio Service Software (RSS). The maximum allowable *known, fixed* differential delay between any two ports that the ASTRO-TAC 3000 comparator can accept (and the RSS allows) is 100 msec (only applies to ASTRO-TAC 3000). The maximum allowable *unknown* differential delay between any two ports that the comparator can accept is 5 msec. (The unknown delay can consist of a combination of *fixed* and *variable* delays.)

If the system delays are not properly optimized, signals from some sites may not be voted by the comparator, even though the signals may have a better signal quality. If improper optimization is allowed to exist, the symptoms may be difficult to later diagnose. Therefore, it is necessary that the optimization procedure be fully understood and followed.

### Automatic Compensation by ASTRO-TAC/ASTRO-TAC 3000 (Digital Systems)

Most small fixed and variable delays can be automatically compensated for by the ASTRO-TAC/ASTRO-TAC 3000. The ASTRO-TAC/ASTRO-TAC 3000 can do this provided that the combination of fixed and variable differential link delay between any two signals does not exceed 5 msec. If a signal exceeds this 5 msec

**Why Simulcast Optimization is Required**

value (without RSS optimization), it is outside the comparator's capability for compensation and may not be considered as a candidate for determining a single best quality or a best quality composite signal. The 5 msec convention is integral to the architecture of the ASTRO-TAC/ASTRO-TAC 3000 and cannot be changed.

It is important to note that the 5-msec convention is a *differential* link delay specification. For example, if one signal arrives at time  $t_1$  and another signal arrives at  $t_2$ , where  $t_2 > t_1 + 5$  msec, the differential delay exceeds the convention and improper operation may result. Since the free-space propagation delay of a radio signal is 5.4  $\mu$ s per mile, an ASTRO subscriber signal has to travel 925 miles before it exceeds the 5 msec ASTRO-TAC/ASTRO-TAC 3000 maximum automatic differential link delay specification. Since ASTRO signals travel slower over Telco facilities, a signal has to travel more than 500 wireline miles (using the worst case copper T1 propagation delay factor) before differential link delay problems are caused by the Telco infrastructure.

**Programmed Differential Link Delay Compensation**

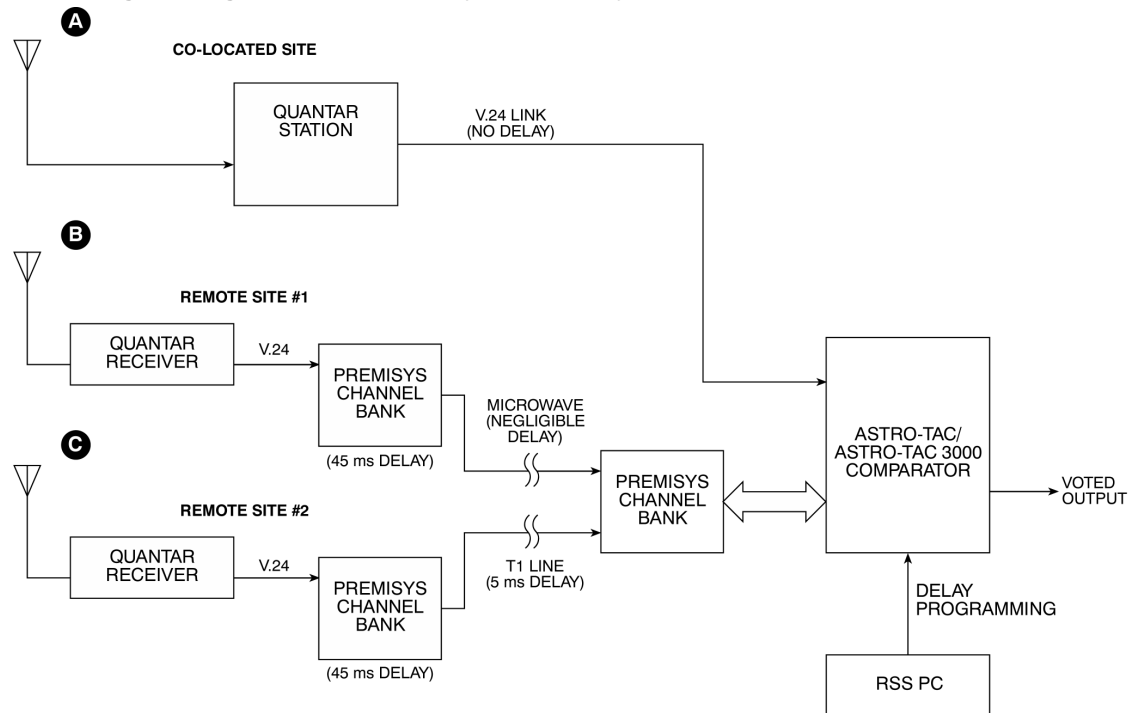
If similar link technologies are used on all links (modems, T1, etc.), the 5 msec automatic differential link delay specification is almost never exceeded, since no *theoretical* differential exists between similar elements. However, delay problems can occur when different technologies are used on different links to the same ASTRO-TAC/ASTRO-TAC 3000 comparator. Figure 7-1 shows an example of such a situation, where a co-located receiver is connected to the ASTRO-TAC/ASTRO-TAC 3000 by a V.24 link and remote receivers connected by a Premisys channel bank. The V.24 link at the co-located site ( **A** in Figure 7-1) has zero delay, but remote site #1 ( **B** in Figure 7-1) has a 45 msec delay introduced by the Premisys, and remote site #2 ( **C** in Figure 7-1) has a 50 msec delay introduced by the Premisys and the T1 link. This known, fixed differential delay is compensated for by the comparator through RSS programming.

In Figure 7-1, the system shown is assumed to be a single region system with customer-owned infrastructure. Because the infrastructure is privately owned, the variability and length of transmission delays can be easily characterized. If the infrastructure delays are undocumented, or if the infrastructure is not customer-owned and controlled, it is recommended that the delays be measured using the procedure described in the "Optimize by V.24 Link Delay Measurement Procedure" section or "Optimize by Analog Link Delay Measurement Using Modems Procedure" section, as applicable.

If *variable* delays exceeding 5 msec are present in the system, the optimization procedure will *not* help. The procedure compensates for *fixed* delays only. The infrastructure must keep variable delays to less than 5 msec.

Both the ASTRO-TAC and the ASTRO-TAC 3000 compensate for differential delays essentially by adding a corresponding amount of delay to each shorter link in order to match the link delay of the link with the greatest delay. However, the procedures in which the delays are programmed using the RSS are somewhat different between the ASTRO-TAC and the ASTRO-TAC 3000.

Link delay programming for both the ASTRO-TAC and the ASTRO-TAC 3000 is individually described in the following section,

**Why Simulcast Optimization is Required****Figure 7-1 RSS Programming for Differential Delays in Mixed System****ASTRO-TAC**

RSS delay programming works by allowing delay to be added to each link.

LINK <b>A</b> DELAY	0 MSEC.
LINK <b>B</b> DELAY	45 MSEC.
LINK <b>C</b> DELAY	50 MSEC.



Using RSS, enter "50 msec" to make this link equal the delay at longest link.



Using RSS, enter "5 msec" to make this link equal the delay at longest link.



This is the longest link. Therefore, using RSS enter "0 msec" for this link.

**ASTRO-TAC 3000**

RSS works by simply entering the total delay of each link into the ASTRO-TAC 3000. ASTRO-TAC 3000 automatically calculates and applies delay to shorter links to match delay of longest link.

ASTRO-TAC 3000 with Ping automatically compensates link delays.

LINK <b>A</b> DELAY	0 MSEC.
LINK <b>B</b> DELAY	45 MSEC.
LINK <b>C</b> DELAY	50 MSEC.



Using RSS, enter "0 msec" for this link.



Using RSS, enter "45 msec" for this link.



Using RSS, enter "50 msec" for this link.

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**Why Simulcast Optimization is Required**

**RSS Link Delay Programming (ASTRO-TAC).** In the system shown in Figure 7-1 where remote satellite receivers are connected by a Premisys channel bank along with a direct V.24 local link, delays would be added as shown in Figure 7-1. Link **C** represents the link with the longest delay. Therefore, the other links must have delay added to make them match link **C**. As such, a delay of 50 msec would be added to the co-located port (link **A**) using RSS to increase that port's delay to match link **C**. Similarly, a delay of 5 msec would be added to the link **B** port using RSS to increase that port's delay to match Link **C**. The ASTRO-TAC software can then process all receiver signals within the 5 msec specification.

**RSS Link Delay Programming (ASTRO-TAC 3000).** In the system shown in Figure 7-1 where remote satellite receivers are connected by a Premisys channel bank along with a direct V.24 local link, the different delay values of the different links would simply be directly entered using RSS. The ASTRO-TAC 3000 software considers all the entered delays and basically compensate the shorter delays (links **A** and **B**) to accordingly match the longest delay; no manual calculations would be required. All receiver signals can then be processed within the 5 msec differential link delay specification.

**RSS Link Delay Programming (ASTRO-TAC 3000 with Ping).** Ping is a handshake between the modems which is used to determine link delay. The ASTRO-TAC 3000 with Ping sends a data frame (ping) out over each link and measures the delay over each link. In the system shown in Figure 7-1 where remote satellite receivers are connected by a Premisys channel bank along with a direct V.24 local link, the different delay values of the different links would *automatically* be measured and compensated by the RSS. The measured delay values for each link are displayed via the RSS. The ASTRO-TAC 3000 software considers all the measured delays and basically compensate the shorter delays (links **A** and **B**) to accordingly match the longest delay; no manual calculations or entering of delay values would be required. All receiver signals can then be processed within the 5 msec specification.

When ping is initiated via the RSS, the following occurs at all wireline ports configured as a Base Station/Receiver. With an output link state at idle (not busy or failed), a ping timer is started and an High-level Data Link Control (HDLC) test command containing a time stamp is sent to the Base Station/Receiver. Upon receipt of the HDLC command the Base Station/Receiver converts the frame into an HDLC test response frame containing the same time stamp. The frame is then sent back to the ASTRO-TAC 3000 (assuming the link is still idle).

Upon receipt of the HDLC test response frame from the Base Station/Receiver, the ASTRO-TAC 3000 then calculates the link delay as follows:

$$(\text{Current time} - \text{Initial time stamp held in HDLC test response frame})/2 = \text{Link delay}$$

The link delay value is then displayed on the RSS. The ping function continues to run until link delays for all links are calculated, or the user exits the RSS local status window. Note that the divide-by-two in the above equation is used to derive link delay for **one** direction, as the ping function naturally returns a value

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**Why Simulcast Optimization is Required**

for a round trip. The premise of simple division to obtain the one-way delay assumes equal propagation delay in **each** direction.

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## Optimization Requirements

### Analog System Optimization

Analog system optimization is accomplished through programming of the DSM-II modules which interface analog channels to the system.

#### When is optimization required?

Perform optimization procedures if:

- ☐ A system is installed or modified (added channels, sites, etc).
- ☐ Degradation of signal quality for any site is noticed.

Perform analog system optimization in accordance with “Analog System Optimization” section of this chapter.

### Digital System Optimization

Both inbound and outbound digital system (ASTRO) optimization is accomplished through RSS programming of the ASTRO-TAC/ASTRO-TAC 3000 comparator.

#### When is optimization required?

Perform optimization procedures if:

- ☐ A system is installed or modified (added channels, sites, etc). Perform inbound and outbound procedures.
- ☐ Degradation of signal quality for any site is noticed. Perform inbound and outbound procedures.
- ☐ The 5 msec maximum differential link delay automatic adjustment range of the ASTRO-TAC/ASTRO-TAC 3000 comparator is exceeded. Perform inbound procedure.

Perform digital system optimization as follows:

- ☐ Inbound Optimization –
  - ☐ Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000) section on page 7-57, **or**
  - ☐ Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping) section on page 7-71
- ☐ Digital (ASTRO) System Outbound Optimization section on page 7-75



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## Analog System Optimization

The following procedure describes how to level-set and optimize an analog (Non-ASTRO) simulcast channel that uses DSM-IIIs and Quantar/Quantro base stations. The DSM-II can be configured for Clear Only, 2L SecureNet, and 4L SecureNet.

Before beginning simulcast optimization, the fixed network equipment must be installed and appropriately configured. Refer to applicable equipment manuals for procedures.

### Prerequisites

Verify the following before starting the optimization procedure.

- ☐ Communication with each remote channel bank is possible from the prime site.
- ☐ Verify all audio paths for each channel in the system.
- ☐ All base stations are PTT-controllable by turning on the E1 Leads at the remote TeNSr channel bank.
- ☐ The optimization consolette(s) is programmed and can receive an adequate RF signal from all remote sites on all channels. Verify the signal quality using an oscilloscope to ensure minimum distortion due to multipath. A rotatable antenna that selects the best signal path is required if multipath distortion is encountered. Multipath distortion generally occurs in mountainous or urban areas with tall buildings.
- ☐ The T1 or E1 system is set up properly in terms of the format, line code, and clocking.

### Required Test Equipment

The following equipment is required for performing the analog optimization procedures:

- ☐ HP3400B RMS Voltmeter (Motorola equipment number 0180302E25) or its equivalent, or a Dynamic Signal Analyzer (HP35665A or HP35670A) with the TTN6110A setup disk. (For either equipment, appropriate rack mounting kits must be ordered.)

<b>NOTE</b>
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This procedure assumes the use of the Dynamic Signal Analyzer (DSA). This piece of equipment is strongly recommended.

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**Analog System Optimization**

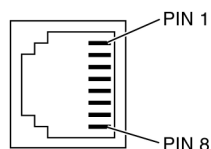
- ☐ Communication Systems Analyzer (service monitor) Motorola R2600 Series or Motorola R2000 Series (with SecureNet option if system is equipped for SecureNet operation).
- ☐ Transmission Test Set with appropriate cables (HP3551A or equivalent).
- ☐ RJ-45 Modular Breakout Adapter (MODAPT).
- ☐ Simulcast Optimization Console (part no. depends on frequency band of system). Appropriate rack mounting kit must be ordered.
- ☐ Laptop PC equipped with the required RSS and VT100 Terminal Emulation Software (PCPLUS or equivalent).
  - ☐ 9600 bps
  - ☐ 8 bits
  - ☐ no parity
- ☐ PC to TeNSr Channel Bank Cable (9-pin D connector-to-RJ48 serial cable), Motorola equipment no. 58D82653X34
- ☐ Oscilloscope (Tektronix 2212 or equivalent). (Motorola equipment no. 0180302E52)

**Recommended Optional Test Equipment**

An RF wattmeter with appropriate cables and adapters may be helpful in certain cases when performing the analog optimization procedures.

**TeNSr Terminal Port Pinout**

Figure 7-2 shows the TeNSr terminal port.



PIN #	FUNCTION	DIRECTION
4	Signal Ground	N/A
5	Receive Data	Output
6	Transmit Data	Input
All other pins are unconnected.		

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Figure 7-2 **TeNSr Terminal Port Pinout**

## Recommended Reference Publications

- ☐ QUANTAR and QUANTRO Service Manual (Motorola manual 68P81095E05)
- ☐ QUANTAR and QUANTRO Station Radio Service Software Guide (Motorola manual 68P81085E35)
- ☐ TeNSr Manual (Motorola manual 68P80309F69)
- ☐ Trunked Radio System Dual Path and Digital Path Simulcast Equipment (Motorola manual 68P81081E60)

## Transmission Test Set Usage

The HP3551 Transmission Test Set (or equivalent) is equipped with a receive load selector switch, which selects the impedance connected across the receive path. The Transmission Test Set loading must be appropriately set to work with the circuits under test. The two Transmission Test Set loading switch positions are terminated and bridged.

- ☐ Terminated: This position is used when the circuit under test is not connected to its  $600\Omega$  load. In this case, the terminated position provides the required  $600\Omega$  termination.
- ☐ Bridged: This position is used when the circuit under test is already terminated and does not require additional termination. In this case, the bridged position provides a high-impedance connection to the circuit under test.

### Line Termination Check (When Measuring a Signal)

The following procedure determines the proper setting of the Transmission Test Set loading selector for a given circuit being tested.

1. Connect Transmission Test Set to desired circuit.
2. While observing dB reading, switch the loading selector from the Receive Bridge position to the Receive Terminate position and note the dB difference between the two positions.
  - 2.1 If the difference between the two positions is 3.5 dB, the circuit under test is terminated. Set selector to Bridge position.
  - 2.2 If the difference between the two positions is near 6 dB, the circuit under test is unterminated. Set selector to Terminate position.

### Line Termination Check (When Generating a Signal)

When using the Transmission Test Set to transmit a signal, the circuit being transmitted into cannot be double-terminated by the circuit's own impedance and the test set termination. (In the transmit mode, the test set always provides a termination). Perform the following procedure to check if double-termination exists, and correct the condition.

**Analog System Optimization**

1. Connect Transmission Test Set to desired circuit.
2. Connect jumpers from the test set transmit +/- audio ports to the receive +/- audio ports.
3. Measure the looped-back audio. Looped-back audio must be equal to the transmit audio  $\pm 0.2$  dBm. If the audio level is off by about 3 dB, the transmission test set is double-terminating the circuit.

If double-termination is noticed, disconnect the circuit feeding the circuit under test.

**Phase Optimization**

The following procedures accomplish the phase optimization for a simulcast system that uses DSM-II modules. Individual procedures are provided for dedicated T1/E1 links, alternative routing (loop-microwave) links, and non-dedicated (leased) T1/E1 links.

**NOTE**

Unlike past versions of simulcast systems, the phase optimization must be performed before the Amplitude Optimization.

If a TeNSr-based simulcast system uses leased T1 lines and an external Channel Service Unit (CSU), the CSU must be configured to not loopback toward the T1 network when there is a failure of the channel bank-to-CSU connection.

**Phase Optimization (Clear Systems)**

Depending on the link type between sites, proceed to the following applicable procedure. Before attempting to perform optimization, verify of the following:

- ☐ All DSM-II ports on the channel must be correctly programmed.
- ☐ Communication links (T1/E1, dial-up line, etc.) between the prime site and all remote site channel banks are required.
- ☐ All GPS receivers are locked to the GPS system.
- ☐ If the communication link for the remote channel bank is over an SRU link, DO NOT place the T1/E1 at the remote channel bank in loopback or run any of the channel bank's internal T1/E1 test from the prime site.

**NOTE**

Do not change the configuration for the remote site SRU port, or run any internal test for the port that is providing the communication link to the remote site from the prime site. Reconfiguring or running internal test from the prime site breaks the communication link to the remote site. If this happens, personnel must go to the remote site to re-establish the link by reconfiguring the T1/E1 or SRU to normal operation through a direct connection to the TeNSr channel bank.

**Dedicated T1/E1 (Microwave or Leased Lines)**

Perform dedicated T1/E1 link optimization as follows:

1. (Trunking only) Disable the channel under test at the prime controller TIB module.
2. Connect the laptop PC serial port to the TERM connector on channel bank interface card.
3. Log into the TeNSr channel bank at the remote site assumed to have the longest infrastructure link delay.
4. From the Main window of the DSM-II for the channel under test, press the <T> key to access the Test window.
5. Place the DSM-II port for the channel under test into test mode as follows:
  - 5.1 Select the desired port by placing the cursor over the TEST field of the desired port. Press the <Enter> key.
  - 5.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 5.3 Press the <S> key to save changes. The port under test is now in the test mode.

**Analog System Optimization**

6. From the Test window, verify the following indications for the DSM-II port:

FRAME SYNC: Yes

1PPS (Pulse Per Second) Present: Yes

MARKER Present: Yes

If the indications above do not exist, proceed as follows:

Error	Corrective Action
No "FRAME SYNC"	<p>Indicates that the DSM-II port is not receiving frame structure from another DSM-II port.</p> <ul style="list-style-type: none"> <li>• Verify that the corresponding port at the prime site is enabled.</li> <li>• Verify that the prime and remote site ports are programmed for the same time slot in the same T1/E1.</li> <li>• Verify that the T1/E1 is up and functioning properly.</li> </ul> <p><b>NOTE:</b> If there is no FRAME SYNC for the remote site, then the MARKER Present field also indicates <b>no</b>.</p>
No "1 PPS Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the remote site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the remote site is connected.</li> <li>• Possible remote site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>
No "MARKER Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the prime site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the prime site is connected.</li> <li>• Possible prime site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>

7. Note and record the value of the excess delay parameter displayed in the Test window EXS Delay field. (Value is expressed in msec.)
8. Repeat steps 3 through 7 for each site in the system.
9. With the excess delays known for each site on the channel, do one of the following:
- 9.1 If the excess delay for each site is at least 2 msec, go to step 11.
- 9.2 If the excess delay is less than 2 msec, or it reports an alarm at any of the sites, then go to step 10.

10. At the site where the excess delay is less than 2 msec (or is displaying an alarm condition), increase the MAX Delay so that the excess delay is at least 2 msec as follows:

**NOTE**

The MAX Delay is expressed in microseconds. The default value is 20000  $\mu$ sec (20 msec).

- 10.1 From the Main window of the DSM-II, move the cursor over the MAX Delay for the port under test and press the <Enter> key.
- 10.2 Enter the new value for the MAX Delay and press the <Enter> key.
- 10.3 Press the <S> key to save this value.
- 10.4 Program the new MAX Delay value in the other remote and loopback or sl-lb ports so that each site has equal launch times. (The MAX Delay is directly proportional to the system throughput.)

**NOTE**

Unless offset equal phase areas are required, the MAX Delay must be equal at each remote site. Refer to "DSM-II Phasing Using 1 PPS" section in Chapter 5 (DSM-II Module) for detailed information.

If the new MAX Delay value is not saved, the old value is still used.

When a DSM-II port is in alarm, the audio output is muted and the E-Leads are released.

11. With MAX Delay set for each site in the system, verify that the internally calculated delay is within 1 to 2  $\mu$ sec of the MAX Delay. The value of the calculated delay is displayed on the Test window in the Delay field.  
If the difference between the MAX Delay and the calculated delay is greater than 2  $\mu$ sec, verify the following:
  - GPS receivers are locked to the GPS system.
  - The T1/E1 performance monitoring is not logging any errors.
  - Clocking for the T1 system is set up properly.
12. (Trunking only) Re-enable the channel under test at Prime Site Controller TIB.

**Analog System Optimization**

13. Repeat steps 1 through 12 for each channel in the system.

**Alternative Routing (Loop-Microwave Systems)**

In a simulcast system that uses alternative routing, there is more than one path out to the remote site. Each of these paths have a different throughput delay. The amount of delay depends upon the distance of the path, as well as the equipment within the path.

Perform alternative routing link optimization as follows:

1. (Trunking only) Disable the channel under test at the Prime Site Controller Transmitter Interface Board (TIB).
2. Estimate which site has the longest path in the system.
3. Estimate the longest path for each alternative route for each site. Of all the estimates, determine the site that has the longest probable route delay.
4. Connect the laptop PC serial port to the TERM connector on the channel bank interface card.
5. Log into the TeNSr channel bank at the remote site with the longest estimated path delay determined in step 3.
6. Configure the microwave system so that the longest path is being used for the remote site being set up (this is either in the clockwise or counter-clockwise direction around the microwave loop).
7. From the Main window of the DSM-II for the channel under test, press the <T> key to access the Test window.
8. Place the DSM-II port for the channel under test into test mode as follows:
  - 8.1 Select the desired port by placing the cursor over the TEST field of the desired port. Press the <Enter> key.
  - 8.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 8.3 Press the <S> key to save changes. The port under test is now in the test mode.



9. From the Test window, verify the following indications for the DSM-II port:

FRAME SYNC: Yes

1PPS (Pulse Per Second) Present: Yes

MARKER Present: Yes

If indications above do not exist, proceed as follows:

Error	Corrective Action
No "FRAME SYNC"	<p>Indicates that the DSM-II port is not receiving frame structure from another DSM-II port.</p> <ul style="list-style-type: none"> <li>• Verify that the corresponding port at the prime site is enabled.</li> <li>• Verify that the prime and remote site ports are programmed for the same time slot in the same T1/E1.</li> <li>• Verify that the T1/E1 is up and functioning properly.</li> </ul> <p><b>NOTE:</b> If there is no FRAME SYNC for the remote site, then the MARKER Present field also indicates <b>no</b>.</p>
No "1 PPS Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the remote site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the remote site is connected.</li> <li>• Possible remote site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>
No "MARKER Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the prime site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the prime site is connected.</li> <li>• Possible prime site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>

10. Note and record the value of the excess delay parameter displayed in Test window EXS Delay field. (Value is expressed in msec.)
11. Repeat steps 6 through 10 for each remaining remote site in the system.
12. With the excess delays known for each site on the channel, do one of the following:
- 12.1 If the excess delay for each site is at least 2 msec, go to step 14.
- 12.2 If the excess delay is less than 2 msec, or it reports an alarm at any of the sites, go to step 13.
13. At the site where the excess delay is less than 2 msec (or is displaying an alarm condition), increase the MAX Delay so that the excess delay is at least 2 msec as follows:

**NOTE**

The MAX Delay is expressed in microseconds. The default value is 20000  $\mu$ sec (20 msec).

- 13.1 From the Main window of the DSM-II, move the cursor over the MAX Delay field for the port under test and press the <Enter> key.
- 13.2 Type in the new value for the MAX Delay and press the <Enter> key.
- 13.3 Press the <S> key to save this value.
- 13.4 Program the new MAX Delay value in the other remote and loopback or sl-lb ports so that each site has equal launch times. (The MAX Delay is directly proportional to the system throughput.)

**NOTE**

Unless offset equal phase areas are required, the MAX Delay must be equal at each remote site. Refer to “DSM-II Phasing Using 1 PPS” section in Chapter 5 (DSM-II Module) for detailed information.

If the new MAX Delay value is not saved, the old value is still used.

When a DSM-II port is in alarm, the audio output is muted and the E-Leads are released.

14. For each site in the system, configure the microwave system for the other loop.

Verify that excess delay is at least 2 msec. If the excess delay is less than 2 msec, or it reports an alarm, increase the MAX Delay until the EXS Delay is at least 2 msec.

**NOTE**

This step is a final verification on the MAX Delay setting. If the excess delay is less than 2 msec, or it reports an alarm, then either the longest path was chosen incorrectly in step 2 or there is a piece of equipment within the path that has a longer delay than what is specified.

15. With MAX Delay set for each site in the system, verify that the internally calculated delay is within 1 to 2  $\mu$ sec of the MAX Delay. The value of the calculated delay is displayed on the Test window in the Delay field.

If the difference between the MAX Delay and the calculated delay is greater than 2  $\mu$ sec, verify the following:

- ☐ GPS receivers are locked to the GPS system.
  - ☐ The T1/E1 performance monitoring is not logging any errors.
  - ☐ Clocking for the T1 system is set up properly.
16. (Trunking only) Re-enable the channel under test at Prime Site Controller TIB.
  17. Repeat steps 1 through 16 for each channel in the system.

### Alternative Routing (Leased T1/E1 Lines)

In a simulcast system that uses alternative routing, there is more than one path out to the remote site. Each of these paths has a different throughput delay. The amount of delay depends upon the distance of the path, as well as the equipment within the path.

Perform non-dedicated leased T1/E1 link optimization as follows:

1. (Trunking only) Disable the channel under test at the prime controller TIB module.
2. From the service provider, obtain the maximum network delay for each site in the system.

<b>NOTE</b>
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If the maximum network delay cannot be obtained from the service provider, obtain maximum network delay by performing "Dedicated T1/E1 (Microwave or Leased Lines)" on page 7-13.

3. Connect the laptop PC serial port to the TERM connector on the channel bank interface card.
4. Log into the channel bank at the remote site with longest network delay.
5. From the Main window of the DSM-II for the channel under test, press the <T> key to access the Test window.
6. Place the DSM-II port for the channel under test into test mode as follows:
  - 6.1 Select the desired port by placing the cursor over the TEST field of the desired port. Press the <Enter> key.

**Analog System Optimization**

- 6.2 At the bottom of the window, on and off options appear. Move the cursor over the **on** option and press the **<Enter>** key.
- 6.3 Press the **<S>** key to save changes. The port under test is now in the test mode.

7. From the Test window, verify the following indications for the DSM-II port:

FRAME SYNC: Yes

1PPS (Pulse Per Second) Present: Yes

MARKER Present: Yes

If indications above do not exist, proceed as follows:

Error	Corrective Action
No "FRAME SYNC"	<p>Indicates that the DSM-II port is not receiving frame structure from another DSM-II port.</p> <ul style="list-style-type: none"> <li>• Verify that the corresponding port at the prime site is enabled.</li> <li>• Verify that the prime and remote site ports are programmed for the same time slot in the same T1/E1.</li> <li>• Verify that the T1/E1 is up and functioning properly.</li> </ul> <p><b>NOTE:</b> If there is no FRAME SYNC for the remote site, then the MARKER Present field also indicates <b>no</b>.</p>
No "1 PPS Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the remote site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the remote site is connected.</li> <li>• Possible remote site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>
No "MARKER Present"	<p>Indicates that the GPS receiver/rubidium 1 PPS output at the prime site is not being received from the GPS receiver.</p> <ul style="list-style-type: none"> <li>• Verify that the GPS receiver/rubidium 1 PPS output at the prime site is connected.</li> <li>• Possible prime site GPS receiver/rubidium or the DSM-II card failure. Replace defective hardware and repeat step.</li> </ul>

8. Considering the DSM-II processing delay of 15 msec, program the MAX Delay so that it is equal to the following:

$$\text{MAX Delay} = 15 \text{ msec} + (\text{maximum network delay}) + 2 \text{ msec}$$

With this new value for the MAX Delay, the EXS Delay should be equal to at least 2 msec.

9. Set MAX Delay field as follows:

<b>NOTE</b>
-------------

The MAX Delay is expressed in  $\mu\text{sec}$ . The default value is 20000  $\mu\text{sec}$  (20 msec).

- 9.1 From the Main window of the DSM-II, move the cursor over the MAX Delay field for the port under test and press the <Enter> key.
- 9.2 Type in the new value for the MAX Delay and press the <Enter> key.
- 9.3 Press the <S> key to save this value.
- 9.4 Program the new MAX Delay value in the other remote and loopback or sl-lb ports so that each site has equal launch times. (The MAX Delay is directly proportional to the system throughput.)

**NOTE**

Unless offset equal phase areas are required, the MAX Delay must be equal at each remote site. Refer to “DSM-II Phasing Using 1 PPS” section in Chapter 5 (DSM-II Module) for detailed information.

If the new MAX Delay value is not saved, the old value is used.

When a DSM-II port is in alarm, the audio output is muted and the E-Leads are released.

10. For each of the remaining remote sites, program the MAX Delay with the same value that was programmed in step 8.

Verify that the port has Frame Sync, 1 PPS Present, and Marker Present. Verify that the EXS Delay is at least 2 msec.

**NOTE**

If the excess delay field reports an alarm, then the actual path delay is greater than the MAX Delay. To correct this, increase the MAX Delay until the alarms disappear and the excess delay reads 2 msec. Program this new MAX Delay in the other remote and loopback ports so that each site has equal launch times. The MAX Delay is directly proportional to the system throughput.

11. With MAX Delay set for each site in the system, verify that the internally calculated delay is within 1 to 2  $\mu$ sec of the MAX Delay. The value of the calculated delay is displayed on the Test window in the Delay field.

If the difference between the MAX Delay and the calculated delay is greater than 2  $\mu$ sec, verify the following:

**Analog System Optimization**

- GPS receivers are locked to the GPS system.
  - The T1/E1 performance monitoring is not logging any errors.
  - Clocking for the T1 system is set up properly.
12. (Trunking only) Re-enable the channel under test at Prime Site Controller TIB.
  13. Repeat steps 1 through 12 for each channel in the system.

**System Level Setting****NOTE**

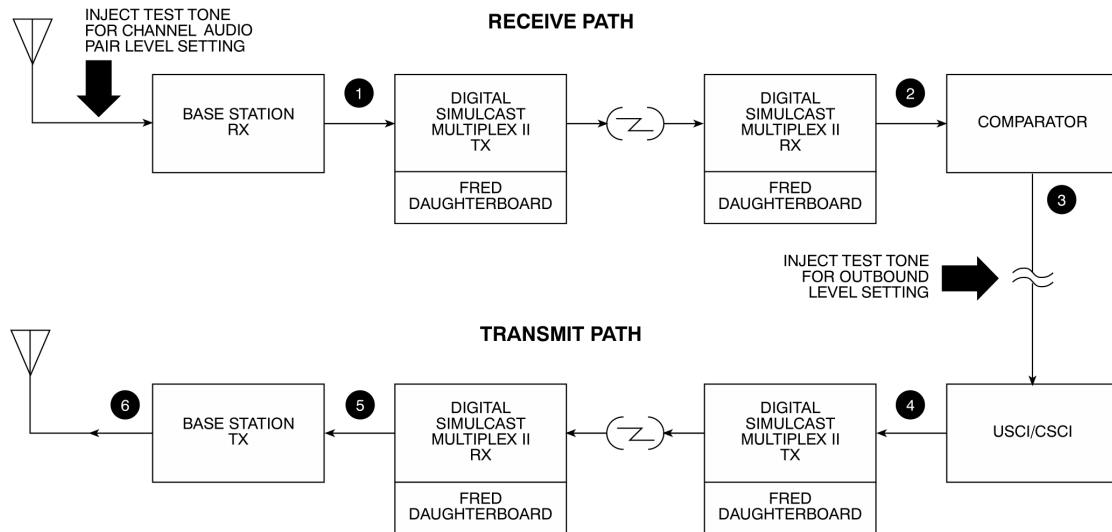
The system performance in a simulcast system depends upon the accuracy of the level setting and optimization procedures. If more than one set of test equipment is used for alignment, you must compare the equipment calibration to verify that they are the same.

Follow the instructions in the exact order given.

The section provides the level setting procedures for the transmit and receive path for an analog simulcast system using DSM-II modules. The transmit path is the block of equipment that begins with the voted audio out of the comparator and ends with the transmitter. The receive path begins at the receiver and ends at the comparator output. Figures 7-3 and 7-4 are system block diagrams (for non-co-located and co-located links, respectively) that show the locations of specific signal levels throughout the system. Signals at these points must be checked and set if required. Tables 7-1 and 7-2 list the corresponding signal levels for the points shown in respective figures 7-3 and 7-4.

System level setting consists of three parts: base station alignment, coarse level adjustment, and fine level adjustment.

## Analog System Optimization



NOTE: FRED DAUGHTERBOARD USED FOR SECURE SYSTEMS ONLY.

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Figure 7-3 **Signal Path Levels Diagram (Non Co-located Links)**

Table 7-1 **Signal Path Levels Table (Non Co-located Links)**

Figure 7-3 Signal Point	Test Point	Required Level
1	DSM-II; Tx audio line (mod in)	-10.0 ±0.5 dBm; 1 kHz tone; ±3 kHz deviation (VHF, UHF, and 806 systems)
		-10.0 ±0.5 dBm; 1 kHz tone; ±2.4 kHz deviation (821 system)
		-10.0 ±0.5 dBm; 1 kHz tone; ±1.5 kHz deviation (896 system)
		-18.6 ±0.5 dBm; status tone (all systems)
		-8.0 ±2.0 dBm; 12 kbit 2L; ±4 kHz deviation (VHF, UHF, and 806 systems)
		-8.0 ±2.0 dBm; 12 kbit 2L; ±2.4 kHz deviation (821 system)
2	DSM-II; Rx audio path (demod out)	-10.0 ±0.5 dBm; 1 kHz tone; ±3 kHz deviation (VHF, UHF, and 806 systems)
		-10.0 ±0.5 dBm; 1 kHz tone; ±2.4 kHz deviation (821 system)
		-10.0 ±0.5 dBm; 1 kHz tone; ±1.5 kHz deviation (896 system)
		-18.6 ±0.5 dBm; status tone (all systems)
		-10.0 ±1.0 dBm; 12 kbit 2L (VHF, UHF, 806, and 821 systems)
3	USCI; bridged audio in	-10.0 ±1.0 dBm; 1 kHz tone; ±3 kHz deviation (VHF, UHF, and 806 systems)
		-10.0 ±1.0 dBm; 1 kHz tone; ±2.4 kHz deviation (821 system)
		-10.0 ±0.5 dBm; 1 kHz tone; ±1.5 kHz deviation (896 system)
		-5.5 ±2.0 dBm; 12 kbit 2L (VHF, UHF, 806, and 821 systems)

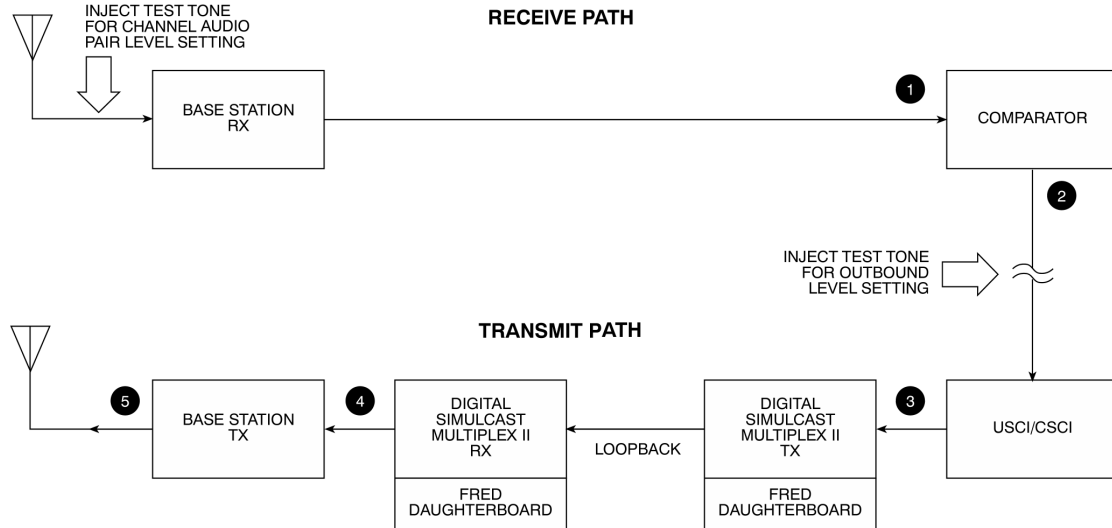
## Analog System Optimization

Table 7-1 **Signal Path Levels Table (Non Co-located Links) (continued)**

Figure 7-3 Signal Point	Test Point	Required Level
4	DSM-II; Tx audio line (mod in)	-10.0 ±0.5 dBm; 1 kHz tone (VHF, UHF, and 806 systems)
		-10.0 ±0.5 dBm; 1 kHz tone (821 system)
		-10.0 ±0.5 dBm; 1 kHz tone (896 system)
		-10.0 ±2.0 dBm; 12 kbit 2L (VHF, UHF, 806, and 821 systems)
5	DSM-II; Rx audio line (demod out)	-10.0 ±0.5 dBm; 1 kHz tone (VHF, UHF, and 806 systems)
		-10.0 ±0.5 dBm; 1 kHz tone (821 system)
		-10.0 ±0.5 dBm; 1 kHz tone (896 system)
		-10.0 ±2.0 dBm; 12 kbit 4L (VHF, UHF, 806, and 821 systems)
6	RF monitor	±3 kHz deviation w/ 1 kHz tone (VHF, UHF, and 806 systems)
		±2.4 kHz deviation w/ 1 kHz tone (821 system)
		±1.5 kHz deviation w/ 1 kHz tone (896 system)
		±1 kHz deviation w/ lowspeed data or PL/DPL (NOTE 2) (VHF, UHF, and 806 systems)
		±800 Hz deviation w/ lowspeed data or PL/DPL (NOTE 2) (821 system)
		±500 Hz deviation w/ lowspeed data or PL/DPL (NOTE 2) (896 system)
		±3 kHz (±200 Hz) deviation w/ 12 kbit 4L (VHF, UHF, and 806 systems)
		±3 kHz (±200 Hz) deviation w/ 12 kbit 4L (821 system)
		±4 kHz (± 200 Hz) deviation w/ 12 kbit 2L (VHF, UHF, and 806 systems)
		±2.4 kHz (± 200 Hz) deviation w/ 12 kbit 2L (821 system)
<b>NOTE:</b> 1. Signal points 1 and 2 are "inbound path." Signal points 3 through 6 are "outbound path." 2. Disconnect word (or PL/DPL) can be measured at USCI/CSCI output; signal level is approximately -19.5 dBm.		



## Analog System Optimization



NOTE: FRED DAUGHTERBOARD USED FOR SECURE SYSTEMS ONLY.

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Figure 7-4 **Signal Path Levels Diagram (Co-located Links)**

Table 7-2 **Signal Path Levels Table (Co-located Links)**

Figure 7-4 Signal Point	Test Point	Required Level
1	Base station; line 2 Rx audio line	-10.0 $\pm$ 0.5 dBm; 1 kHz tone; $\pm$ 3 kHz deviation (VHF, UHF, and 806 systems)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone; $\pm$ 2.4 kHz deviation (821 system)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone; $\pm$ 1.5 kHz deviation (896 system)
		-18.6 $\pm$ 0.5 dBm; status tone (all systems)
		-8.0 $\pm$ 2.0 dBm; 12 kbit 2L; $\pm$ 4 kHz deviation (VHF, UHF, and 806 systems)
		-8.0 $\pm$ 2.0 dBm; 12 kbit 2L; $\pm$ 2.4 kHz deviation (821 system)
2	USCI; bridged audio in	-10.0 $\pm$ 1.0 dBm; 1 kHz tone; $\pm$ 3 kHz deviation (VHF, UHF, and 806 systems)
		-10.0 $\pm$ 1.0 dBm; 1 kHz tone; $\pm$ 2.4 kHz deviation (821 system)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone; $\pm$ 1.5 kHz deviation (896 system)
		-5.5 $\pm$ 2.0 dBm; 12 kbit 2L (VHF, UHF, 806, and 821 systems)
3	DSM-II; Tx audio line (mod in)	-10.0 $\pm$ 0.5 dBm; 1 kHz tone (VHF, UHF, and 806 systems)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone (821 system)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone (896 system)
		-10.0 $\pm$ 2.0 dBm; 12 kbit 2L (VHF, UHF, 806, and 821 systems)
4	DSM-II; Rx audio line (demod out)	-10.0 $\pm$ 0.5 dBm; 1 kHz tone (VHF, UHF, and 806 systems)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone (821 system)
		-10.0 $\pm$ 0.5 dBm; 1 kHz tone (896 system)
		-10.0 $\pm$ 2.0 dBm; 12 kbit 4L (VHF, UHF, 806, and 821 systems)

## Analog System Optimization

Table 7-2 **Signal Path Levels Table (Co-located Links) (continued)**

Figure 7-4 Signal Point	Test Point	Required Level
5	RF monitor	±3 kHz deviation w/ 1 kHz tone (VHF, UHF, and 806 systems)
		±2.4 kHz deviation w/ 1 kHz tone (821 system)
		±1.5 kHz deviation w/ 1 kHz tone (896 system)
		±1 kHz deviation w/ lowspeed data or PL/DPL (NOTE 2) (VHF, UHF, and 806 systems)
		±800 Hz deviation w/ lowspeed data or PL/DPL (NOTE 2) (821 system)
		±500 Hz deviation w/ lowspeed data or PL/DPL (NOTE 2) (896 system)
		±3 kHz (±200 Hz) deviation w/ 12 kbit 4L (VHF, UHF, and 806 systems)
		±3 kHz (±200 Hz) deviation w/ 12 kbit 4L (821 system)
		±4 kHz (± 200 Hz) deviation w/ 12 kbit 2L (VHF, UHF, and 806 systems)
		±2.4 kHz (± 200 Hz) deviation w/ 12 kbit 2L (821 system)
<b>NOTE:</b> 1. Signal point 1 is “inbound path.” Signal points 2 through 5 are “outbound path.” 2. Disconnect word (or PL/DPL) can be measured at USCI/CSCI output; signal level is approximately -19.5 dBm.		

**Base Station Alignment (Quantar™/Quantro®)**

Perform base station alignment as described in the following section.

**NOTE**

Perform this procedure for each base station in the system.

The following procedure applies only to Quantar/Quantro base stations. If the Simulcast system uses the MSF5000/10000 base station, refer to "Trunked Radio System Dual Path and Digital Path Simulcast Equipment" (Motorola manual 68P81081E60).

1. (Trunking only) Disable the trunked channel being aligned on the Prime Central Controller (TIB).
2. Read the base station's code plug using Quantar/Quantro Radio Service Software (RSS).
3. Enter Service Mode on RSS.
4. From the **service:alignment** menu, press <F4> to access station disable.
5. Perform base station checks and adjustments in accordance with Table 7-3. (For a detailed base station alignment procedure, refer to the reference publication listed in table.)

Table 7-3 **Base Station Alignment**

Characteristic	Procedure	Reference Publication
Reference Oscillator	<ul style="list-style-type: none"> <li>Verify Quantar/Quantro base station internal reference oscillator is netted to 5 MHz input from GPS rubidium frequency standard.</li> </ul> <p><b>NOTE:</b> This alignment takes a minimum of 5 minutes per station. If the station control board is changed, the autonet alignment must be performed again.</p> <p><b>NOTE:</b> You can use the manual procedure in place of the auto-net procedure to save time. The Reference Oscillator alignment is required, even for a system using the external frequency reference, however the manual procedure is sufficiently accurate for this adjustment.</p>	QUANTAR and QUANTRO Station Radio Service Software Guide (Motorola manual 68P81085E35)
TX Power Output	<ul style="list-style-type: none"> <li>Verify output power; adjust if required.</li> </ul> <p><b>NOTE:</b> TX power output is typically pre-aligned at the factory.</p>	
Squelch	<ul style="list-style-type: none"> <li>Set squelch according to system design (portable coverage, mobile coverage, etc.) to ensure proper voting operation.</li> </ul>	
Receive Wireline	<ul style="list-style-type: none"> <li>Verify receive wireline is set to produce a 1 kHz tone @ -10 dBm when a 1 kHz tone @ 60% system deviation on channel carrier is present.</li> <li>Verify status tone level is set to -18.6 dBm.</li> </ul> <p><b>NOTE:</b> The status tone is based on the peak audio level. For a -10 dBm output level, the peak audio is set to -5.6 dBm. Therefore, the number entered for the Status Tone field in the RSS is -13. <math>(-5.6 + (-13) = -18.6)</math></p>	
TX Deviation Gain Adjustment	<ul style="list-style-type: none"> <li>Verify base station is set for 60% system deviation.</li> </ul> <p><b>NOTE:</b> TX deviation is aligned at the factory.</p> <p><b>NOTE:</b> If the alignment is off by <math>\pm 100</math> Hz (or less), this is compensated for externally during Coarse Level Adjustment.</p>	
Reference Modulation Compensation	<ul style="list-style-type: none"> <li>Verify that modulation compensation (MOD COMP) is adjusted by going through appropriate procedure. If MOD COMP is properly adjusted, then make no changes.</li> </ul> <p><b>NOTE:</b> There is no room for error in this adjustment. Error of only one step off causes audio degradation in non-capture/overlap areas. For procedure information about setting MODCOMP, see the "Modulation Compensation" section.</p>	

## Modulation Compensation

The information describing how to adjust modulation compensation includes the following topics:

- ☐ Rough Setting
- ☐ Simulcast Setting
- ☐ Fine-Tuning Modulation Compensation

### **Rough Setting**

When adjusting the modulation compensation using the procedure found in the QUANTAR and QUANTRO Station Radio Service Software Guide (Motorola manual 68P81085E35), the Communications System Analyzer (CSA) must have the low pass and high pass filters set properly.

If you are using the R2600 CSA, do the following:

1. Press the **<SPF>** key.
2. Use the cursor arrows to highlight the **High Pass** filter option.
3. Verify that the filter is set to 5 Hz.
4. Use the cursor arrows to highlight the **Low Pass** filter option.
5. Verify that the setting for the filter is 300 Hz.

If you are using the R2000 CSA, do the following:

1. On the numeric keypad, simultaneously press both arrow keys.
2. Select the Special function for the receive filters.
3. Enable the 5 Hz highpass filter and the 300 Hz low pass filter.

Certain earlier R2000-series CSAs may not have the Special function option. This is verified if the special function menu is not displayed.

### **Simulcast Setting**

Modulation compensation is critical to overall system optimization. Since the goal is to achieve less than a 0.3 dB difference in amplitude response between sites, the shape of the low frequency end of the Frequency response curve is important. The modulation compensation adjustment determines this shape.

The procedure for aligning Reference Modulation Compensation in the QUANTAR and QUANTRO Station Radio Service Software Guide (Motorola manual 68P81085E35) results in an alignment that is accurate for most purposes, but not accurate enough for Simulcast. Additional fine-tuning of the modulation compensation adjustment is required. You must use a Dynamic Signal Analyzer (DSA) (HP35665A or HP35670A) to verify that the frequency response curve is as flat as possible.

When using the DSA to view the frequency response, it becomes apparent that the step size of the modulation compensation adjustment on some base stations may not be fine enough to achieve a flat frequency response. On these stations, you can flatten the frequency response by adjusting the transmitter deviation using the technique described in the “Fine-Tuning Modulation Compensation” section.

Due to the simulcast equipment configuration, this additional fine-tuning step is easier to perform as part of the Fine Amplitude Adjustment procedure. It is also possible to make this adjustment during initial Base Station Alignment if the DSA and the optimization receiver are available when aligning the modulation compensation.

### **Fine-Tuning Modulation Compensation**

The purpose of the modulation compensation (mod-comp) adjustment is to flatten the frequency response of the base station at low frequencies by adjusting the gain at low frequencies. A DSA is required to display the frequency response from 1 to 100 Hz when setting the modulation compensation. Figure 7-5 shows the frequency response of the base station from 1 to 100 Hz at several different mod-comp settings. The response curves are arbitrarily referenced to zero at 100 Hz.

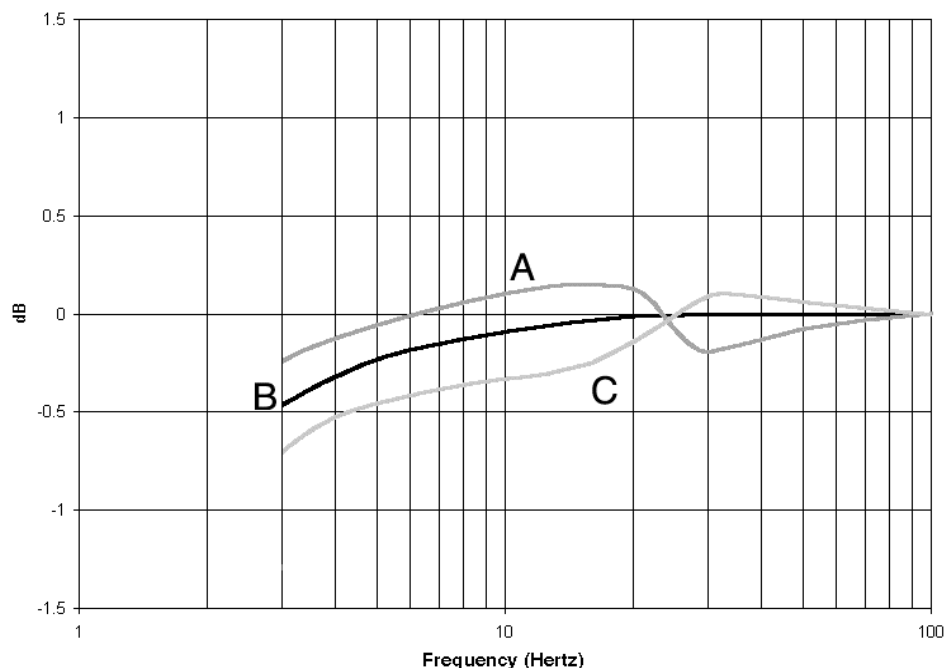


Figure 7-5 **Frequency Response**

The goal is to adjust the modulation compensation to achieve a flat response, as shown by trace B. If the modulation compensation is set high, the frequency response looks like trace A. If the modulation compensation is low, the response looks like trace C. Ideally the response depicted by trace B should be attainable by the adjustment of the modulation compensation. However, sometimes this is not possible because of the step size of the modulation compensation adjustment. Please note that in Figure 7-5, trace A and trace C are about 0.4 dB apart. This is the nominal step size of the modulation compensation adjustment on some base stations. Therefore, trace A and C represent the worst case where one setting is too high and the next lower setting is too low.

When this situation occurs, it still is possible to obtain a flat frequency response (trace B) by making a small adjustment in the gain at high frequencies. This is possible because the step size in the adjustment of the high frequency audio gain is about 0.1 dB. Trace B can be obtained by selecting the modulation compensation setting that gave you trace A and increasing the deviation slightly to boost the high frequency levels, thereby flattening the frequency response. In

**Analog System Optimization**

order to adjust the high frequency level, increase the deviation by modifying the TX Deviation Adjustment using the RSS.

If you lower the numbers that you entered in the TX Deviation Gain Adjust window when you did the initial station alignment, the deviation becomes higher. Lowering the entered deviation values by about 40 Hz should result in a 0.1 dB increase in deviation. Due to variations in the stations, a 40 Hz change may not always be sufficient. If a change does not occur, increase the value slightly.

Figure 7-5 shows the worst case situation where the desired frequency response (trace B) is exactly halfway between two modulation compensation settings. In this case, if you started with trace A, you would lower the numbers in the TX Deviation Gain Adjust window by 80 Hz to increase the deviation by the 0.2 dB needed to flatten the frequency response. This makes trace A look exactly like trace B except that it is 0.2 dB above trace B. The gain in the channel bank is then reduced by 0.2 dB when performing the Fine Amplitude Adjustment to move trace A down to match trace B.

In the worst case, you should only need to change the deviation by 0.2 dB. Use the modulation compensation setting to get as close as possible to the desired frequency response, then make the smallest possible change to the deviation setting necessary to achieve a flat frequency response. Changing the deviation by altering the data entered in the RSS deviation window affects the analog and the ASTRO deviation. Though changing the ASTRO deviation less than 0.2 dB has no measurable effect on system performance, avoid larger changes in deviation.

To save time, change only two of the values on the TX Deviation Gain Adjust window. The TX Deviation Gain Adjust window prompts you to enter the deviation that you measure at four different test frequencies. These frequencies represent the end points of the two VCO ranges. In Simulcast, the base station operates only on one frequency so that you only need to adjust two of the fields. Determine the normal operating frequency of the base station and identify which two of the four test frequencies that it falls between. You only need to modify these two deviation settings.

**NOTE**

If you set the modulation compensation by using the steps in section “Combined Channel Alignment Procedure,” you do not have to perform the procedures in sections “Fine Amplitude Adjustment (Clear Systems Optimization)” and “Final System Level Check.”

**Combined Channel Alignment Procedure**

In this procedure, after displaying the frequency response of a channel at the reference site from 1 to 100 Hertz with a vertical sensitivity of at least 0.5 dB per division, verify that the modulation compensation is set properly (the frequency response is flat) and store this trace as a reference to use when setting the rest of the channels. Then superimpose each additional site’s frequency response over the reference site and adjust the modulation compensation to obtain the flattest

frequency response. Finally, adjust the gain on the channel bank until the traces overlap.

By using the following procedure with a Dynamic Signal Analyzer (DSA) (HP35665A or HP35670A), you set the gain, check the modulation compensation adjustment, and perform the final system level check all in one operation.

1. Perform steps 5, 6, 7, and 9 of the procedure in section “Fine Amplitude Adjustment (Clear Systems Optimization).”
2. Connect a MODAPT between the USCI and SDA (or DSMII if Broadcast Mode).
3. Connect the SOURCE output of the DSA to the MODAPT pins 4 and 5.
4. On the USCI close S1-1 (Mute Data) and S1-2 (Mute Audio).
5. To connect the consolette to the DSA, do the following:
  - 5.1 Connect consolette discriminator cable to Channel 1 input on the DSA.
  - 5.2 Select the appropriate channel on the consolette.
6. To set up the DSA, do the following:
  - 6.1 On the DSA, press the Source button.
  - 6.2 Select **Source On**.
  - 6.3 Select **Periodic Chirp**.
  - 6.4 Select **Level**.
  - 6.5 Set the level to 241 mVrms.
  - 6.6 Press the **Window** button.
  - 6.7 Select **Uniform**.
  - 6.8 Press the **Freq** (frequency) button.
  - 6.9 Select **Start**.
  - 6.10 Set Start frequency to 1 Hz.
  - 6.11 Select **Stop**.
  - 6.12 Set Stop frequency to 100 Hz.
  - 6.13 Press the Trace Coord button.
  - 6.14 Select **X-Axis - Log**.
7. To capture a Trace, do the following:
  - 7.1 Press the **Scale** button.
  - 7.2 Select **Autoscale On**.
  - 7.3 Press the **Start** button.
  - 7.4 Press the **Pause - Cont** (continue) button when you have acquired one trace.
  - 7.5 Select **Autoscale Off**.

**Analog System Optimization**

- 7.6 Select **Center Reference**.
- 7.7 Adjust the marker knob to center the trace.
- 7.8 Select **Y per Div** (Decades).
- 7.9 Set to 0.5 dB. (If the signal is noisy, try averaging a number of traces.)
- 8. To save the trace as a reference in D1, do the following:
  - 8.1 Press the **Save/Recall** button.
  - 8.1 Select **Save Data**.
  - 8.2 Select **Save Trace**.
  - 8.3 Select **Into D1**.

The reference trace is saved in D1.

- 9. Display D1 as a reference trace.
  - 9.1 Press the **Active Trace** button to change to the second available trace. (In addition, on the HP35670A DSA, select B.)
  - 9.2 To display the reference trace:
    - A. Press the **Meas Data** (measure data) button.
    - B. Select **More**.
    - C. Select **Data Register**.
    - D. Select **D1**.
  - 9.3 Press the **Active Trace** button to return to the live trace. (In addition, on the HP35670A DSA, select A.)
  - 9.4 To display both traces:
    - A. Press the **Disp Format** button.
    - B. Select **Front/Back**.

The reference trace appears on the window. You can now compare data taken from the other sites with the reference site.

- 10. Set up to monitor the next site.
- 11. Adjust the modulation compensation to obtain a flat trace.
- 12. Adjust the gain using the channel bank so that the overlay does not deviate from the reference by more than 0.3 dB.
- 13. Perform steps 10, 11, and 12 for the same channel at all remote sites.



## ASTRO-TAC 3000 Wireline Alignment

In addition to the other procedures specified in this section, perform the following procedure for systems using ASTRO-TAC 3000.

### NOTE

The following procedure must be performed for each ASTRO-TAC 3000 in the system.

1. (Trunking only) Disable the trunked channel that is being aligned on the Prime Central Controller (TIB).
2. Connect the laptop PC serial port to the terminal connector on the ASTRO-TAC 3000 being aligned.
3. Read the code plug of the ASTRO-TAC 3000 using the RSS.
4. After the code plug is completely read into the service terminal, go to the **Main Menu**.
5. From the **Main Menu**, press the <F4> key to access the Change/Create/View window.
6. From the menu, press the <F5> key to access the Port Control Parameters window.
7. Noting that each wireline port has a number, enter the port number to be aligned.
8. In the Port Type field, select one of the following options using the up and down arrow keys:
  - ☐ Select **AEB** for ports interfacing to an AEB (Trunked Comparator only)
  - ☐ Select **STATION/RECEIVER** for ports interfacing to a station/receiver, USCI, CSCI (Trunked or Conventional Comparator)
  - ☐ Select **DIU** for ports interfacing to a DIU (Conventional Comparator)
9. In the Link Type field, select the HYBRID LINK-EXTERNAL TRANSMIT CLOCK option using the up/down arrow keys.
10. In the Input Level Differential field, enter +3 dB.
11. In the Output Level field, enter the following for each field:
  - ☐ ANALOG: -10 dB
  - ☐ IDLE TONE: -10 dB
12. Repeat steps 7 through 11 for each port that interfaces to the comparator.
13. Press the <F10> key to access **Change/View Codeplug** menu.

## Analog System Optimization

14. Press the <F10> key to go back to **Main Menu**.
15. Press the <F3> key to access **Get/Save/Change Codeplug** menu.
16. Press the <F8> key to save the RSS setup and to program the ASTRO-TAC 3000.
17. Exit the RSS program and repeat steps 1 through 16 for any additional ASTRO-TAC 3000 channels.

## Coarse Level Adjustment (Receive Audio Path)

**NOTE**

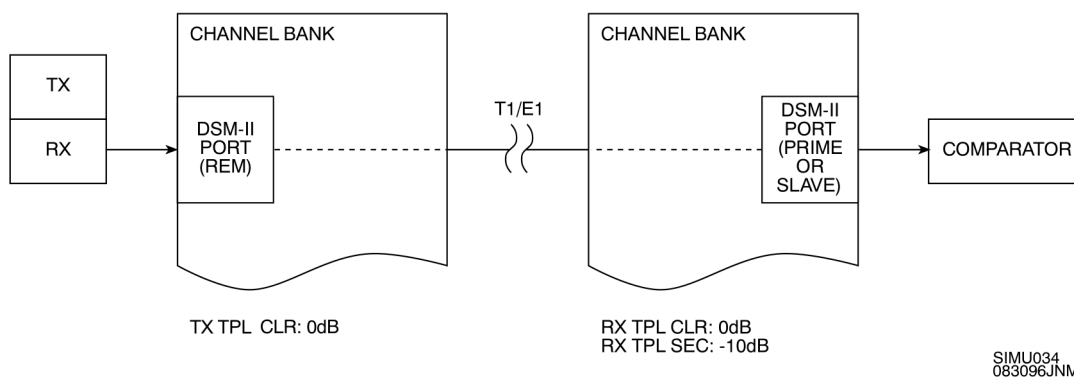
Before attempting to perform coarse level adjustment, make certain all the base stations have been aligned.

Since the TLPs in the DSM-II module are calibrated, the clear transmit TLP at the remote site DSM-II port (rem port) and the clear receive TLP at the prime site DSM-II port (prime or slave port) must set to their default settings of 0 dB. This provides a unity gain clear receive audio path from the receiver to the comparator. Figure 7-6 shows the required receive audio path levels throughout the path.

**NOTE**

The Tx TLP CLR acts as an attenuator and the Rx TLP CLR acts as an amplifier. The RX TLP SEC must be set to its default value of -10 dBm. This provides a SecureNet level of -10 dBm into the comparator.

Figure 7-6 **Receive Audio Path**



## Coarse Level Adjustment (Transmit Audio Path)

### NOTE

Before attempting to perform coarse level adjustment, make certain all the base stations have been aligned.

The transmit audio path starts at the comparator output and ends at the Base Station. On the transmit audio path, the comparator and USCI must be coarse-adjusted to provide a -10 dBm output level @ 1 kHz.

The base station's internal Tx Deviation Adjustment procedure adjusts the transmitter deviation to 60% system  $\pm 100$  Hz. The input level must then be adjusted to eliminate the  $\pm 100$  Hz discrepancy in the deviation. The following procedures describe these adjustments. Figures 7-4 and 7-7 show the required transmit audio path levels throughout the path.

Depending upon the comparator used, perform the following applicable procedure.

### **Coarse Level Adjustment (Comparator other than ASTRO-TAC 3000).**

Perform transmit coarse level adjustment for systems using other than ASTRO-TAC 3000 comparator as follows:

1. On either the prime or slave DSM-II port interfacing to the comparator, turn on the 1 kHz test tone. (With the Rx TLP CLR set to 0 dB, the level of the tone is -10 dBm  $\pm 0.1$  dB.) Activate 1 kHz test tone as follows:
  - 1.1 From the DSM-II Test window, move the cursor with the arrow keys to highlight the TONE field of the desired port. Press the <Enter> key.
  - 1.2 Use the arrow keys to select the 1 kHz tone. When the desired setting is displayed, press the <Enter> key.
  - 1.3 Press the <S> key to save changes. The port outputs a 1 kHz tone at the Rx output.

With the 1 kHz tone going into the comparator, the comparator is unquelled and voting that channel.

2. With the Transmission Test Set in the bridged mode, monitor the audio line between the comparator and the USCI (or CSCI).
3. Adjust the comparator line level out (LD5) fed to the USCI (or CSCI) for a -10 dBm output.
4. On USCI (or CSCI), mute TDATA as follows:
  - On USCI, disable Low Speed Data (Disconnect Word) by closing SW1-1 (Mute Data).
  - On CSCI, disable PL/DPL by closing SW100-2 (Mute PL/DPL).

TDATA mute is verified by green LED on USCI (or CSCI) changing from illuminated to blinking.

**Analog System Optimization**

5. Monitor the audio lines between the USCI (or CSCI) and SDA.

**NOTE**

If the system is configured for broadcast, the monitoring point is between the USCI (or CSCI) and the DSM-II.

6. Adjust the USCI output (R110) or CSCI output (R112) for a -10 dBm output.
7. At each site on the channel, key up the base station and measure the deviation with a service monitor.

**NOTE**

When measuring deviation, verify that received waveform has no noise riding on it. Any noise riding on waveform causes a false reading.

The base station can be keyed-up by activating the E1-Lead in the remote, loopback, or sl-lb DSM-II ports.

8. Verify that the received deviation is at 60% system deviation (1.5 kHz for 896 systems, 2.4 kHz for 821 systems, or 3.0 kHz deviation for VHF, UHF, and 806 systems). Refer to Table 7-4 for deviation settings.

If the deviation is not at 60%, adjust the Rx TLP CLR in the remote site DSM-II port.

9. Repeat steps 7 and 8 for every site on the channel.
10. When the deviation for the last site has been verified, turn off the 1 kHz tone.
11. Repeat steps 1 through 10 for every channel in the system.

**Coarse Level Adjustment (ASTRO-TAC 3000 Comparator).** Perform transmit coarse level adjustment for systems using an ASTRO-TAC 3000 comparator as follows:

1. Disconnect the audio path between the ASTRO-TAC 3000 and the USCI (or CSCI).
2. On USCI (or CSCI), mute TDATA as follows:
  - On USCI, disable Low Speed Data (Disconnect Word) by closing SW1-1 (Mute Data).
  - On CSCI, disable PL/DPL by closing SW100-2 (Mute PL/DPL).TDATA mute is verified by green LED on USCI (or CSCI) changing from illuminated to blinking.
3. Using a Transmission Test Set, inject a 1 kHz, -10 dBm tone into the USCI (or CSCI) input.

<b>NOTE</b>
-------------

Refer to applicable system documentation for USCI (CSCI) input pinout location.

4. Monitor the audio lines between the USCI (or CSCI) and SDA.

<b>NOTE</b>
-------------

If the system is configured for broadcast, the monitoring point is between the USCI (or CSCI) and the DSM-II.

5. Adjust the USCI output (R110) or CSCI output (R112) for a -10 dBm output.
6. At each site on the channel, key up the base station and measure the deviation with a service monitor.

<b>NOTE</b>
-------------

When measuring deviation, verify that received waveform has no noise riding on it. Any noise riding on waveform causes a false reading.

The base station can be keyed-up by activating the E1-Lead in the remote, loopback, or sl-lb DSM-II ports.

**Analog System Optimization**

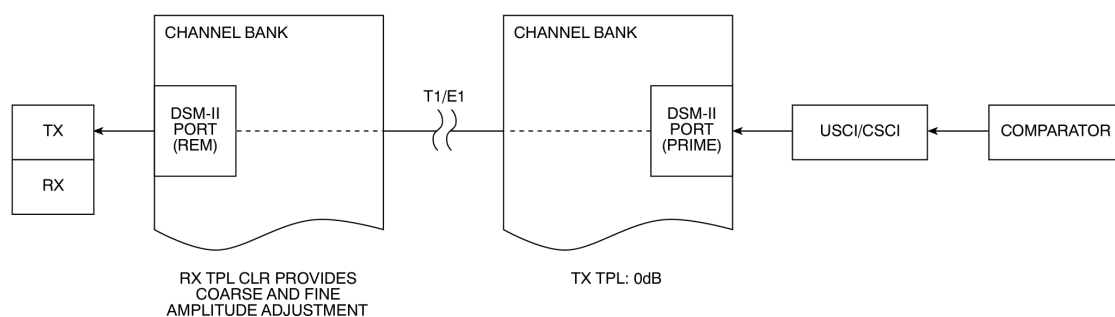
7. Verify that the received deviation is at 60% system deviation (1.5 kHz for 896 systems, 2.4 kHz for 821 systems, or 3.0 kHz deviation for VHF, UHF, and 806 systems). Refer to Table 7-4 for deviation settings.

If the deviation is not at 60%, adjust the Rx TLP CLR in the remote site DSM-II port.

8. Repeat steps 6 and 7 for every site on the channel.
9. When the deviation for the last site has been verified, disconnect the Transmission Test Set and reconnect the ASTRO-TAC 3000 to the USCI (or CSCI) input.
10. Repeat steps 1 through 9 for every channel in the system.

Table 7-4 **Quantar/Quantro Deviation Settings**

Deviation Adjustment	Frequency Range		
	VHF, UHF, and 806-821 MHz	821-824 MHz	896-902 MHz
100% full system deviation	5.0 kHz	4.0 kHz	2.5 kHz
Maximum station deviation	4.6 kHz	3.7 kHz	2.3 kHz
60% full system deviation	3.0 kHz	2.4 kHz	1.5 kHz
40% full system deviation	2.0 kHz	1.6 kHz	1.0 kHz
Trunking data deviation (disconnect word)	1.0 kHz	0.80 kHz	0.50 kHz
Failsoft data deviation	1.0 kHz	0.80 kHz	0.50 kHz
Coded deviation	4.0 kHz	2.4 kHz	none



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Figure 7-7 **Transmit Audio Path**

### Coarse Level Adjustment (Trunked Systems Lowspeed Data Adjustment)

<b>NOTE</b>
-------------

Before attempting to perform coarse level adjustment, make certain all the base stations have been aligned.

Perform lowspeed data coarse level adjustment as follows:

1. Disconnect the audio lines between the comparator and USCI.
2. Enable the channel for trunking operation.
3. With the system in the trunked mode, key up a radio and verify that it is assigned to the channel under test.
4. With a service monitor, measure the deviation of the received lowspeed data. (With the audio lines disconnected, there should be no audio present in the received modulation.)
5. Adjust the lowspeed data at the USCI (R143) for 20% system deviation. (500 Hz for 896 systems, 800 Hz for 821 systems, or 1 kHz for VHF, UHF and 806 systems.)

### Coarse Level Adjustment (Conventional Systems PL/DPL Adjustment)

<b>NOTE</b>
-------------

Before attempting to perform coarse level adjustment, make certain all the base stations have been aligned.

Perform PL/DPL coarse level adjustment as follows:

1. Disconnect the audio lines between the comparator and CSCI.
2. Key up a radio and verify that it is assigned to the channel under test.
3. With a service monitor, measure the deviation of the received PL/DPL. (With the audio lines disconnected, there should be no audio present in the received modulation.)
4. Adjust the PL/DPL data at the CSCI (R141) for 20% system deviation. (500 Hz for 896 systems, 800 Hz for 821 systems, or 1 kHz for VHF, UHF and 806 systems.)

**Analog System Optimization****Fine Amplitude Adjustment (Clear Systems Optimization)**

This procedure fine tunes the DSM-II output so the modulation levels (deviation) of all transmitters on a channel are equal for audio.

The following test equipment is required to perform this procedure.

- ☐ HP3400A RMS Voltmeter, or a DSA (HP35665A or HP35670A) with the TTN6110A setup disk.
- ☐ Motorola R2001, R2024, or R2600 CSA (service monitor)
- ☐ Simulcast Optimization Console (Syntor or Spectra)

Perform fine amplitude adjustment as follows:

1. Set up test equipment as shown in Figure 7-8.
2. (Trunking only) Disable the channel under test at the Prime Site Controller TIB.
3. Using an oscilloscope, monitor the recovered audio to determine the quality of the recovered audio. The recovered audio must be a clean sine wave without multipath distortion.



4. Set the optimization consolette to monitor the channel under test. Verify that there are no active carriers on the channel.

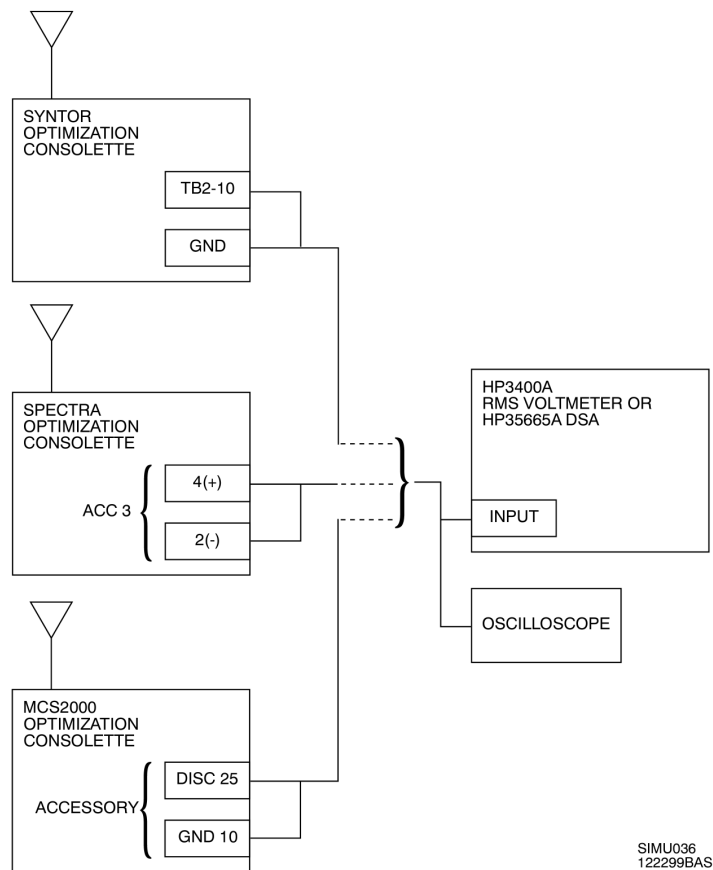


Figure 7-8 **Amplitude Optimization Test Setup**

5. Log into the TeNSr channel bank of the co-located remote site (or the remote site nearest to the prime site). (This site serves as the reference site.)
6. From the Main window of the DSM-II with the reference port, press the <T> key to access Test window.

#### NOTE

The reference port is the DSM-II port at the reference site that is directly connected to the base station. The reference port is either a rem, loopback, or sl-lb DSM-II port.

7. Place the DSM-II reference port for the channel under test into test mode as follows:
  - 7.1 Select the desired port by placing cursor over the TEST field of the desired port. Press the <Enter> key.

**Analog System Optimization**

- 7.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the **<Enter>** key.
    - 7.3 Press the **<S>** key to save changes. The port under test is now in the test mode.
  8. From the Test window of the reference port, turn on the internal 20 Hz tone as follows:
    - 8.1 From the Test window, move the cursor with the arrow keys to highlight the TONE field of the reference port. Press the **<Enter>** key.
    - 8.2 At the bottom of the window, **off**, **1 khz**, **20 hz**, and **cal** options appear. Move the cursor over the **20 hz** option and press the **<Enter>** key.
    - 8.3 Press the **<S>** key to save this change. The port outputs a 20 Hz tone at the Rx output.
  9. From the Test window of the reference port, turn on the E1-Lead to issue a PTT to the base station as follows:
    - 9.1 From the Test window, use the arrow keys to highlight the E1 field of the reference port. Press the **<Enter>** key.
    - 9.2 At the bottom of the window, **norm**, **on**, and **off** options appear. Move the cursor over the **on** option and press the **<Enter>** key.
    - 9.3 Press the **<S>** key to save changes. The E1-Lead is enabled for this port.
  10. If you are using the DSA, perform the following steps:
    - 10.1 Verify that the DSA has been configured using the TTN6110A setup disk and that channel 2 of the DSA is connected to the console's discriminator output.
    - 10.2 Use the [ACTIVE TRACE] HARDKEY in the DISPLAY section of the DSA to select Trace A. A 20 Hz signal should be present on the X-axis of the display.

The Marker (diamond shaped) should be present at the peak of the signal. The current frequency (X) and amplitude (Y) values are located above the trace.

<b>NOTE</b>
-------------

If the peak does not appear on the display or extends beyond the limitations of the display, adjust the TOP REFERENCE level in the DISPLAY section under **SCALE** menu. Press the **<F2>** key to display the current value. If the display disappears, do not turn the knob because it changes the location of the marker.

- 10.3** Press the [MARKER] HARDKEY in the MARKER section of the DSA. Press REFERENCE TO MARKER to set up the current values as reference values (square shaped marker). The Xr (frequency) and Yr (amplitude) values are now relative to the reference value. This reference value is used to match the deviations of the transmitters. (On the HP35670A, however, REFERENCE TO MARKER is a selection in the **REFERENCE SETUP** softkey menu.)
- 11.** If you are using the HP3400A RMS Voltmeter, measure and record the voltage output (Vref in dB). This reference value is used to match the deviations of the transmitters on the channel.

**NOTE**

Record the reference value obtained in step 10.3 (or 11), as it is used later.

- 12.** From the Test window, turn off the internal 20 Hz tone and place the E1-Lead back to normal operation. Turn off the test mode for the reference DSM-II port.
- 13.** Select and log into a different remote site TeNSR channel bank.
- 14.** From the Main window of the DSM-II with the selected port, press the <T> key to enter the Test window.
- 15.** Place the DSM-II port for the channel under test into test mode as follows:
- 15.1** Select the desired port by placing cursor over the TEST field of the desired port. Press the <Enter> key.
  - 15.2** At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 15.3** Press the <S> key to save changes. The port under test is in test mode.
- 16.** From the Test window of the selected port, turn on the internal 20 Hz tone as follows:
- 16.1** From the Test window, move the cursor with the arrow keys to highlight the TONE field of the selected port. Press the <Enter> key.
  - 16.2** At the bottom of the window, **off**, **1 khz**, **20 hz**, and **cal** options appear. Move the cursor over the **20 hz** option and press the <Enter> key.
  - 16.3** Press the <S> key to save this change. The port outputs a 20 Hz tone at the Rx output.
- 17.** From the Test window of the selected port, turn on the E1-Lead to issue a PTT to the base station as follows:
- 17.1** From the Test window, use the arrow keys to highlight the E1 field of the selected port. Press the <Enter> key.
  - 17.2** At the bottom of the window, **norm**, **on**, and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.

**Analog System Optimization**

- 17.3 Press the <S> key to save changes. The E1-Lead is enabled for this port.
18. As applicable, use the DSA or HP3400A RMS Voltmeter to measure the signal at the consolette discriminator output.
19. Note the reference value obtained in step 10.3 (or 11). Adjust the RX TPL CLR of the selected port until the consolette's discriminator output is equal to the reference value as follows:
- 19.1 In the DSM-II Card window for the selected port, use the arrow keys to highlight the Rx TPL CLR field corresponding to the selected port.
- 19.2 When the desired field is highlighted, press the <Enter> key. An entry box appears at the bottom of the window.
- 19.3 Enter a value to adjust the TPL and press the <Enter> key.

**NOTE**

RX TPL CLR range is -16.3 dB to +6.8 dB in 0.1 dB increments.

- 19.4 Press the <S> key to save the change.
- 19.5 Repeat steps 19.1 through 19.4 until consolette discriminator output value matches reference value.
20. From the Test window, turn the E1-Lead back to the normal position and turn off the internal 20 Hz tone. Turn off the test mode.
21. Repeat steps 13 through 20 for each site on the channel.
22. Repeat steps 2 through 21 for each channel in the system.

**Final System Level Check****NOTE**

Final System Level Check must be performed for every channel in the system.

Motorola R2600-Series CSA, or equivalent equipment is capable of generating a test frequency of lower than 20 Hz, must be used. The Transmission Test Set does not work for this test.

**NOTE**

If a DSA was used to align the system, you have already performed this System Level Check as part of the Combined Channel Alignment Procedure.

If a DSA is not available, this procedure allows you to plot the frequency response of the system using a voltmeter.

The following test equipment is required to perform this procedure.

- ☐ HP3400A RMS Voltmeter
- ☐ Motorola R2600 or R2000 CSA (service monitor)
- ☐ Simulcast Optimization Console (Syntor or Spectra)

Perform final system level check as follows:

1. (Trunking only) Disable the channel under test at the Prime Site Controller TIB.
2. Perform the test setup as shown in Figure 7-9. On the Service Monitor, connect the MOD OUT connector to the DVM COUNTER connector using a BNC-to-BNC cable.

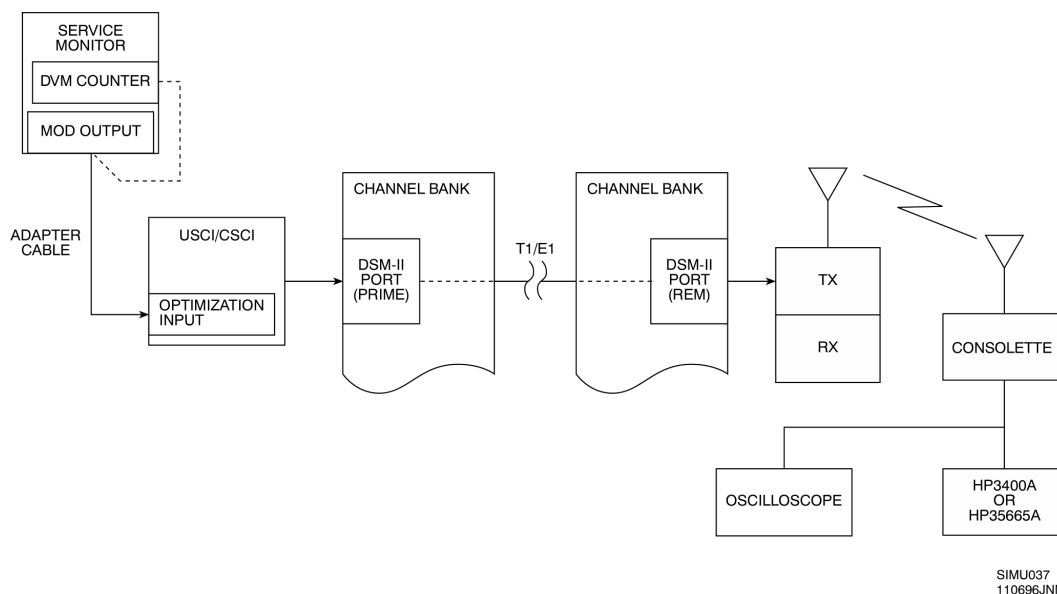


Figure 7-9 **Final System Check Test Setup**

## Analog System Optimization

3. On USCI, disable Low Speed Data (Disconnect Word) by closing SW1-1 (Mute Data). On CSCI, disable PL/DPL by closing SW100-2 (Mute PL/DPL). the green LED on USCI or CSCI changes from illuminated to blinking.
4. Set R2600 or R2000 CSA to generate 20 Hz tone as described in the following steps.

4.1 If you are using an R2600 Series CSA, set controls as follows:

Switch or Control	Position or Setting
RF control	MONITOR position
FORMAT SEL switch	TONE A position (NOTE)
FREQ switch	20 Hz/ 15 Hz/ 12 Hz/ 10 Hz/ 7 Hz/ 5 Hz
Meter	AC voltmeter
SYNTH control	Adjust level for approximate -10 dBm output at USCI/CSCI (check level with DVM/counter on CSA).
<b>NOTE:</b> The tone must be continuous. The setting is displayed just to the right of the SYNTH level field. x indicates tone is off. ~ indicates continuous tone.	

4.2 If you are using an R2000 Series CSA, set controls as follows:

Switch or Control	Position or Setting
FUNCTION switch	GENERATE position
DISPLAY	DVM/DIST position
MODULATION control	CONT. position
CODE SYNTH MODE switch	PL/DPL position
PL FREQ switch	20 Hz/15 Hz/12 Hz/10 Hz/7 Hz/5 Hz
CODE SYNTH control	Adjust level for approximate -10 dBm output at USCI/CSCI (check level with DVM/counter on CSA).

5. (See Figure 7-9.) Use the BNC-to-Bantam cable shipped with the system to connect the MOD OUT connector on service monitor to USCI or CSCI.
6. Log into the TeNSr channel bank of the station under test.
7. From the Main window of the DSM-II channel under test, press the <T> key to enter the Test window.

8. Place the DSM-II port for the channel under test into test mode as follows:
  - 8.1 Select the desired port by placing cursor over the TEST field of the desired port. Press the <Enter> key.
  - 8.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 8.3 Press the <S> key to save changes. The port under test is now in the test mode.
9. For DSM-II port feeding the remote base station (DSM-II programmed as rem, loopback, or sl-lb), key up base station as follows:
  - 9.1 From the Test window, use the arrow keys to highlight the E1 field of the selected port. Press the <Enter> key.
  - 9.2 At the bottom of the window, **norm**, **on**, and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 9.3 Press the <S> key to save changes. The E1-Lead is enabled for this port.

**NOTE**

In the following steps, the reference value recorded during the “Fine Amplitude Adjustment (Clear Systems Optimization)” procedure (step 10.3 or 11) is required as a reference.

10. Make several photocopies of Table 7-5. This table serves as a log for recovered signal levels at various test frequencies. The data entered is used to check and verify system alignment.

**NOTE**

In steps 11 through 18, verify that the output of the USCI/CSCI and the consolette exhibit a clean sine wave. Also verify that the modulation is approximately 60%.

11. Observe the reading on the HP3400A RMS Voltmeter.
12. Using an oscilloscope, monitor the recovered audio to determine the quality of the recovered audio. The recovered audio must be a clean sine wave without multipath distortion.
13. In the copy of Table 7-5 that is used for your system log, enter the 20 Hz reference level obtained in “Fine Amplitude Adjustment (Clear Systems Optimization)” procedure.
14. Set the frequency of the generator to 15 Hz. Note the level recovered from the Optimization Consolette. Enter this level in the 15 Hz column of Table 7-5 for the site.

**Analog System Optimization**

15. Set the frequency of the generator to 12 Hz. Note the level recovered from the Optimization Console. Enter this level in the 12 Hz column of Table 7-5.
16. Set the frequency of the generator to 10 Hz. Note the level recovered from the Optimization Console. Enter this level in the 10 Hz column of Table 7-5 for the site.
17. Set the frequency of the generator to 7 Hz. Note the level recovered from the Optimization Console. Enter this level in the 7 Hz column of Table 7-5 for the site.
18. Set the frequency of the generator to 5 Hz. Note the level recovered from the Optimization Console. Enter this level in the 5 Hz column of Table 7-5 for the site.
19. Set to E1-lead that was enabled in step 9 to norm. Key up the next base station at the next site as described in step 9.
20. Repeat steps 6 through 19 for all sites in the system.
21. Re-enable the channel just checked and disable the next channel to be measured.
22. Repeat steps 1 through 21 for all channels in the system.

*Table 7-5 Recovered Level Entry Log*

CHANNEL # ____						
Site:	20 Hz:	15 Hz:	12 Hz:	10 Hz:	7 Hz:	5 Hz:
Site 1						
Site 2						
Site 3						
Site 4						
Site 5						
Site 6						
Site 7						
Site 8						
Site 9						
Site 10						



**NOTE**

The output of the CSA begins to drop when the frequency generated falls below 10 Hz. This causes the roll-off seen in the chart and is acceptable.

23. Verify that for each given frequency, the levels recorded in Table 7-5 are no more than 0.3 dB down for each site at a given frequency.

If the levels between sites for a given frequency vary by more than 0.3 dB, this indicates a misaligned Reference Modulation Compensation on one or more Base Stations. Repeat “Base Station Alignment (Quantar™/Quantro®)” procedure (located on page 7-26) for any base station exceeding 0.3 dB misalignment. MOD COMP waveform, as displayed on CSA, should appear as a square wave with rising edge that transitions into a basically flat upper level with no humps or dips.

A hump in the MOD COMP waveform indicates that the low frequency (5 Hz, 7 Hz, 10 Hz) levels are hotter than the levels at the higher frequencies (12 Hz, 15 Hz, 20 Hz). In this case, decrease the MOD COMP level.

A dip in the MOD COMP waveform indicates that the low frequency (5 Hz, 7 Hz, 10 Hz) levels are decreasing too rapidly relative to the levels at the higher frequencies (12 Hz, 15 Hz, 20 Hz). In this case, increase the MOD COMP level.

**NOTE**

Moving the Reference Modulation Compensation by only one step on the Quantar/Quantro RSS causes approximately a 0.4 dB change at 5 Hz. For this reason, setting the initial MOD COMP is extremely critical to overall system optimization. A 0.3 dB difference between sites causes significant noise in Non-Capture/Overlap areas.

## Fine Tuning (SECURENET® (12 kbit) Systems)

### Phase Optimization (SECURENET® (12 kbit) Systems)

The phase optimization procedure for the SecureNet systems is the same as the clear systems. Performing the clear phase optimization automatically covers the SecureNet phase optimization.

### Amplitude Optimization (SECURENET® (12 kbit) Systems)

This procedure fine tunes the DSM-II output so the data modulation levels (deviation) of all transmitters on a channel are equal.

The following test equipment is required to perform this procedure.

- ☐ HP3400A RMS Voltmeter, or a DSA (HP35665A or HP35670A) with the TTN6110A setup disk.
- ☐ Motorola R2001, R2024, or R2600 CSA (service monitor)
- ☐ Simulcast Optimization Consolette:
  - ☐ 800 MHz Spectra - L35KXM7174AM
  - ☐ 900 MHz Spectra - L27KMM7170AM

Perform final system level check as follows:

1. Set up test equipment as shown in Figure 7-8.
2. (Trunking only) Disable the channel under test at the Prime Site Controller TIB.
3. Log into the TeNSr channel bank at the prime site that contains the prime port for the channel under test.

<b>NOTE</b>
-------------

When using the DSM-II in a SecureNet simulcast system, the broadcast configuration must be used. In the broadcast configuration, there is only one prime port per channel.

4. From the Main window of the DSM-II channel under test, press the <T> key to enter the Test window.
5. Place the DSM-II port for the channel under test into test mode as follows:
  - 5.1 Select the desired port by placing cursor over the TEST field of the desired port. Press the <Enter> key.
  - 5.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.

- 5.3 Press the **<S>** key to save changes. The port under test is now in the test mode.
6. From the Test window of the selected port, turn on the internal 1.5 kHz (sqr) squarewave as follows:
  - 6.1 On the prime port of the desired channel, highlight the SEC OPT field. Press the **<Enter>** key.
  - 6.2 Use the arrow keys to highlight the SEC OPT field. Press the **<Enter>** key.
  - 6.3 At the bottom of the window, the **off**, **sqr**, **stair**, and **random** options appear. Move the cursor over the **sqr** option and press the **<Enter>** key.
  - 6.4 Press the **<S>** key to save changes. The 1.5 kHz squarewave is enabled.
7. From the Main window of the DSM-II that contains the reference port for the channel under test, press the **<T>** key to enter the Test window.

**NOTE**

If the reference port is in a different channel bank from the prime DSM-II port, then log into the channel bank containing the reference port.

8. Place the DSM-II port for the channel under test into test mode as follows:

**NOTE**

This port is either a sl-lb or a rem port.

- 8.1 Select the desired port by placing cursor over the TEST field of the desired port. Press the **<Enter>** key.
- 8.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the **<Enter>** key.
- 8.3 Press the **<S>** key to save changes. The port under test is now in the test mode.
9. From the Test window of the reference port, turn on the E1-Lead to issue a PTT to the Base Station as follows:
  - 9.1 From the Test window, use the arrow keys to highlight the E1 field of the selected port. Press the **<Enter>** key.
  - 9.2 At the bottom of the window, **norm**, **on**, and **off** options appear. Move the cursor over the **on** option and press the **<Enter>** key.
  - 9.3 Press the **<S>** key to save changes. The E1-Lead is enabled for this port.

## Analog System Optimization

10. Use the CSA to measure the deviation at the reference site.

**NOTE**

Set CSA to wide band mode.

11. At the reference port, adjust the Rx TLP SEC so that the deviation is 3 kHz. Adjust the Rx TLP SEC as follows:
  - 11.1 Using the arrow keys, highlight the Rx TPL SEC field corresponding to the reference port.
  - 11.2 When field is highlighted, press **<Enter>**. An entry box appears at the bottom of the window.
  - 11.3 Enter a value to adjust the TPL. Press the **<Enter>** key when done.

**NOTE**

RX TPL SEC range is -12 dB to -4 dB in 0.1 dB increments.

- 11.4 Press the **<S>** key to save the change.
  - 11.5 Repeat steps 11.1 through 11.4 until 3 kHz deviation is verified by CSA.
12. Set the optimization consolette to monitor the channel under test.
13. If you are using the DSA, perform the following steps:
  - 13.1 Verify that the DSA has been configured using the TTN6110A setup disk and that channel 2 of the DSA is connected to the consolette's discriminator output.
  - 13.2 Use the [ACTIVE TRACE] HARDKEY in the DISPLAY section of the DSA to select Trace A. A 1.5 kHz signal should be present on the X-axis of the display.

The Marker (diamond shaped) should be present at the peak of the signal. The current frequency (X) and amplitude (Y) values are located above the trace.

**NOTE**

If the peak does not appear on the display or extends beyond the limitations of the display, adjust the TOP REFERENCE level in the DISPLAY section under **SCALE** menu. Press **<F2>** to display the current value. If the display disappears, do not turn the knob because it changes the location of the marker.

- 13.3 Press the [MARKER] HARDKEY in the MARKER section of the DSA. Press REFERENCE TO MARKER to set up the current values as reference values (square shaped marker). The Xr (frequency) and Yr (amplitude) values are now relative to the reference value. This reference value is used to match the deviations of the transmitters. (On the HP35670A, however, REFERENCE TO MARKER is a selection in the **REFERENCE SETUP** softkey menu.)
14. If you are using the HP3400A RMS Voltmeter, measure and record the voltage output (Vref in dB). This reference value is used to match the deviations of the transmitters. The reference site must be keyed with the 1.5 kHz squarewave.

**NOTE**

Record the reference value obtained in step 13.3 (or 14), as it is used later.

15. From the Test window, place the E1-Lead back to normal operation. Turn off the test mode for the reference DSM-II port.
16. Select and log into a different remote site TeNSR channel bank.
17. From the Main window of the DSM-II with the selected port, press the <T> key to enter the Test window.
18. Place the DSM-II port for the channel under test into test mode as follows:
- 18.1 Select the desired port by placing cursor over the TEST field of the desired port. Press the <Enter> key.
  - 18.2 At the bottom of the window, **on** and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 18.3 Press the <S> key to save changes. The port under test is in test mode.
19. From the Test window of the selected port, turn on the E1-Lead to issue a PTT to the base station as follows:
- 19.1 From the Test window, use the arrow keys to highlight the E1 field of the selected port. Press the <Enter> key.
  - 19.2 At the bottom of the window, **norm**, **on**, and **off** options appear. Move the cursor over the **on** option and press the <Enter> key.
  - 19.3 Press the <S> key to save changes. The E1-Lead is enabled for this port.
20. As applicable, use the HP35665A DSA or HP3400A RMS Voltmeter to measure signal at the console discriminator output.

**Analog System Optimization**

21. Note the reference value obtained in step 13.3 (or 14). Adjust the RX TPL SEC of the selected port until the console's discriminator output is equal to the reference value as follows:
    - 21.1 In the DSM-II Card window for the selected port, use the arrow keys to highlight the Rx TPL SEC field corresponding to the selected port.
    - 21.2 When the desired field is highlighted, press the **<Enter>** key. An entry box appears at the bottom of the window.
    - 21.3 Enter a value to adjust the TPL and press the **<Enter>** key.
- NOTE**
- RX TPL SEC range is -12 dB to -4 dB in 0.1 dB increments.
- 21.4 Press the **<S>** key to save the change.
  - 21.5 Repeat steps 24.1 through 24.4 until console discriminator output value matches reference value.
22. From the Test window, turn the E1-Lead back to the normal position. Turn off the test mode.
  23. Repeat steps 16 through 22 for each site on the channel.
  24. Repeat steps 2 through 23 for each channel in the system.

**Data Polarity Check**

A simulcast system requires all transmitter modulations to be identical. In addition, the trunking data, conventional data (DPL), and SecureNet Data (4L FRED) is polarity sensitive. For proper simulcast system operation, all audio connections must be correct with the proper polarization (positive to positive and negative to negative) to all remote sites.

Incorrect polarity causes the following problems:

- ☐ Distorted audio in the overlap area
- ☐ RF data issues, for example, subscribers are not able to decode data such as control channel, lowspeed data, DPL, and 12kbit (FRED)
- ☐ ReSC not being able to lock to control channel (CRB lost of lock error)

In a conventional system, the sub-audible tones/data is in the form of either PL or DPL. PL is not polarity sensitive when in capture of a site. PL is only polarity sensitive in the overlap area. DPL is polarity sensitive in either the overlap or in capture of a site. When the channel is configured for PL during normal operation, it must be temporarily configured for DPL to perform the polarity check.

Perform polarity inversion check and correction for trunking systems or conventional systems as described in the following sections.

## Trunking Systems Polarity Check/Correction

Perform over-the-air trunking systems polarity check/correction in accordance with Table 7-6.

Table 7-6 **Polarity Check/Correction Procedure (Trunking Systems)**

Step	Procedure	Indication	Corrective Action
<p><b>NOTE:</b></p> <p>Before performing procedure, make certain of the following:</p> <ul style="list-style-type: none"> <li>• Base station is operating properly with VSWR properly set.</li> <li>• Low-speed data deviation is set to 20% of system deviation.</li> <li>• Channel under test does not have any interfering carriers.</li> <li>• Properly programmed subscriber radio (correct system ID and connect tone) is available to conduct test.</li> </ul>			
1	Verify that the Prime Central Controller (PCC) assigns each control channel.	PCC assigns each control channel. Go to step 2.	
		PCC does not assign each control channel and all ReSCs show a CRB out of lock error.	Check for polarity inversion between USCI and SDA (non-broadcast mode channel) or USCI and DSM-II (broadcast modes channel).
		ReSC cannot lock to the RTIB TDATA input.	<p>Check the following:</p> <ul style="list-style-type: none"> <li>• Make certain J4 on RTIB is set to invert.</li> <li>• (Quantar/Quantro) Make certain RCV Signal Inversion parameter is disabled.</li> </ul> <p>Check for polarity inversion between the SDA and DSM-II (non-broadcast mode channel) or DSM-II and Base Station (broadcast mode and non-broadcast mode channels).</p> <p>This condition is further indicated by the DISABLE LED being illuminated on the ReSC board with no channel control LEDs illuminated on IRB.</p>
2	Select a voice-only channel. Verify Rx subscriber can stay on the voice channel while in capture of each site.	Rx subscriber stays on channel. End of procedure.	
		Subscriber cannot hold the voice channel while in capture of any site.	Check for polarity inversion between the USCI and SDA (non-broadcast mode channel) or USCI and DSM-II (broadcast modes channel).
		Subscriber cannot hold the voice channel while in capture of a particular site.	Check for polarity inversion between the SDA and DSM-II (non-broadcast mode channel) or DSM-II and Base Station (broadcast modes channel).

## Analog System Optimization

**Conventional Systems Polarity Check/Correction**

Perform over-the-air trunking systems polarity check/correction in accordance with Table 7-7.

Table 7-7 **Polarity Check/Correction Procedure (Conventional Systems)**

Step	Procedure	Indication	Corrective Action
<b>NOTE:</b> Before performing procedure, make certain of the following: <ul style="list-style-type: none"> <li>• Base station is operating properly with VSWR properly set.</li> <li>• Low-speed data deviation is set to 20% of system deviation.</li> <li>• Channel under test does not have any interfering carriers.</li> <li>• Properly programmed subscriber radio (correct DPL code) is available to conduct test.</li> <li>• When the channel is configured for PL during normal operation, it must be temporarily configured for DPL to perform the polarity check.</li> </ul>			
1	Key up a subscriber while in capture of each site.		
2	Verify Rx subscriber can stay on the voice channel while in capture of each site.	Rx subscriber stays on channel. End of procedure.	
		Subscriber cannot hold the voice channel while in capture of any site.	Check for polarity inversion between the CSCI and SDA (non-broadcast mode channel) or CSCI and DSM-II (broadcast modes channel).
		Subscriber cannot hold the voice channel while in capture of a particular site.	Check for polarity inversion between the SDA and DSM-II (non-broadcast mode channel) or DSM-II and Base Station (broadcast modes channel).

**NOTE**

Only the subscriber's side and the CSCI have to be reconfigured for DPL. The base station does not require reprogramming.



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## Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)

<b>NOTE</b>
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This section applies only to the ASTRO-TAC and earlier ASTRO-TAC 3000 comparators **without** the ping feature. For ASTRO-TAC 3000 comparator with ping feature, go to “Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping)” section.

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Depending on the link type used, there are three procedures for predicting or measuring the link delay:

- ☐ Optimize By Calculating Delay From System Diagrams
- ☐ Optimize By V.24 Link Delay Measurement
- ☐ Optimize By Analog Link Delay Measurement Using Modems

In the Measurement and Optimization portions of each procedure, different procedures are provided for the ASTRO-TAC and ASTRO-TAC 3000 comparators.

Depending on the link type(s) and comparator used, perform the appropriate inbound optimization procedure shown in the following sections.

### Optimize By Calculating Delay From System Diagrams Procedure

<b>NOTE</b>
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This method is recommended only for privately owned infrastructure where the signal routing and delays are documented and known to be constant.

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The easiest method of calculating the link delays for each receiver link is to use the system diagrams and known delays of system components. Depending upon the comparator type, optimize links as described in the following sections.

## Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)

**Measurement and Optimization (ASTRO-TAC)**

1. Separately calculate and record the delay for each link as follows:  
**(link length x delay value) + equipment delays**

**NOTE**

Premisys SRU must be configured with delay optimization set to **on**.

2. If the delay between any two links (determined in step 1) exceeds 5 msec, calculate the delay value to be added that makes the link(s) with shorter delays have a delay equal to the link with the greatest delay (targeting a zero differential delay).
3. Using RSS, enter the values determined in step 2 using the field: MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG:LINK DELAY.

Table 7-8 lists the typical delay values for various transmission mediums. Table 7-9 lists the typical delay values for link equipment used in Motorola installations.

**Measurement and Optimization (ASTRO-TAC 3000)**

1. Separately calculate and record the delay for each link as follows:  
**(link length x delay value) + equipment delays**

**NOTE**

Premisys SRU must be configured with delay optimization set to **on**.

2. For each link delay determined in step 1, directly enter the delay value using the RSS field: MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG:LINK DELAY.

Table 7-8 lists the typical delay values for various transmission mediums. Table 7-9 lists the typical delay values for link equipment used in Motorola installations.

*Table 7-8 Typical Transmission Medium Delays*

Medium Type	Typical Delay Value
Copper wire T1	8-10 $\mu$ s/mile
Fiber Optic T1	10 $\mu$ s/mile
Microwave T1	7 $\mu$ s/mile
Co-located Digital Cross-connect Switch (CO DAX)	125-500 $\mu$ s/switch

Table 7-9 **Link Delay Values For Typical Motorola Installations**

Link Type	End-to-End Unidirectional Delay (Typical)
Motorola ASTRO WL modem	12 msec/link
Premisys Sub-Rate Unit (SRU) sync 9.6	45 msec/link (NOTE 1)
Siemens Data Service Unit (DSU) sync 9.6	3-6 msec/link
UDS V.3225 sync 9.6 trellis	28 msec/link
UDS V.3225 sync 9.6 uncoded	23 msec/link
Quantar analog RF link	20-30 msec/link
<b>NOTES:</b> 1. If SRU contains V2.6 firmware, delay is 22.5 msec/link instead of 45 msec/link. 2. The delays listed in this table are representative. For delays of specific equipment, consult manufacturer's specifications.	

### Example Using System Diagram Delay Optimization (ASTRO-TAC)

Assume a system has an ASTRO-only voting system that uses one co-located base station and two remotely located satellite receivers. There are 10 wireline miles from the ASTRO-TAC to site 1. Copper T1 is used. There are 17 microwave miles to site N. Straight V.24 is used on the co-located base station. Premisys is used on the remote sites. The following delays are optimized using an ASTRO-TAC comparator as listed in Table 7-10.

As seen in the Total Delay column, the co-located site has zero delay, while remote sites Site 1 and Site N both have approximately 45 msec of delay. Therefore, a link delay compensation of 45 msec (Add  $\Delta$  column in Table 7-10) must be added to the co-located port to bring the ports to within 5 msec of differential delay. The Add  $\Delta$  value would be the delay value entered in the comparator RSS to make the co-located port match the other ports.

This delay value would be entered in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG:LINK DELAY field for the co-located port.

Table 7-10 **Delay Calculations Using Diagram Delay Optimization (ASTRO-TAC)**

Link No.	Wireline Miles	Telco Delay	Equipment Delay	Total Delay	Add $\Delta$
Site 1	10	0.1 msec	45 msec	45.1 msec	0 msec
Site N	17	0.09 msec	45 msec	45.09 msec	0 msec
Co-loc	0	0 msec	0 msec	0 msec	45 msec

This example also shows that the Telco contribution to link delay is insignificant. This is usually the case, but not always. Some less-common types of transmission equipment, such as SONET (Synchronous Optical Network), may cause the Telco delay to be larger than expected.

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)****Example Using System Diagram Delay Optimization (ASTRO-TAC 3000)**

Assume a system has an ASTRO-only voting system that uses one co-located base station and two remotely located satellite receivers. There are 10 wireline miles from the ASTRO-TAC to site 1. Copper T1 is used. There are 17 microwave miles to site N. Straight V.24 is used on the co-located base station. Premisys is used on the remote sites. The following delays are optimized using an ASTRO-TAC 3000 comparator as listed in Table 7-11.

As seen in the Total Delay column, the co-located site has zero delay, while remote sites Site 1 and Site N both have approximately 45 msec of delay. With the ASTRO-TAC 3000, the various delay values of each link are directly entered into the comparator. For site 1 and site N, 45 msec would be entered. For the co-located site, 0 msec would be entered.

These delay values would be entered in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG:LINK DELAY field for each port.

*Table 7-11 Delay Calculations Using Diagram Delay Optimization (ASTRO-TAC 3000)*

Link No.	Wireline Miles	Telco Delay	Equipment Delay	Total Delay
Site 1	10	0.1 msec	45 msec	45.1 msec
Site N	17	0.09 msec	45 msec	45.09 msec
Co-loc	0	0 msec	0 msec	0 msec

This example also shows that the Telco contribution to link delay is insignificant. This is usually the case, but not always. Some less-common types of transmission equipment, such as SONET (Synchronous Optical Network), may cause the Telco delay to be larger than expected.

## Optimize by V.24 Link Delay Measurement Procedure

The total link delay on synchronous V.24 links requires optimization if the infrastructure delay is very high compared with that of the Motorola equipment. These delay readings can then be measured and used to determine the differential link delay values to program into the various comparator ports.

### Equipment Required

PN/Description	Manufacturer
Firebird 6000 Telecommunications Test Set	Telecommunications Techniques Corp.

### Setup

(See Figure 7-10.) Perform V.24 measurement setup as follows:

1. Disconnect the ASTRO-TAC/ASTRO-TAC 3000 from the prime site transmission equipment.
2. Connect the transmission equipment V.24 port to the Firebird rear panel V.24 connector. Make sure that the V.24 clock line is connected. (Clock is expected to be sourced by the transmission equipment.)
3. Disconnect the corresponding port at the remote site transmission equipment from the connected station.
4. At remote site transmission equipment, jumper the tx and rx data pins on the V.24 cable in a loop-back configuration.

#### NOTE

For accurate measurement, it is important that an external hardwire loopback be used rather than any software loopback feature that may be available on the remote equipment.

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)****5. Turn on the Firebird and select:**


---

```

<Menu>
    Intf Setup
    <INT232>
    <EMLATE>
    <DTE>
    Test Inverval
    <10E4>
    Char Format
    (not required; just press <enter>)
Auxiliary (Aux Functions)
    <3> <1> <enter>
    Delay
    <start> <more> <more>
    <gpatt> <up arrow> <stop>
    <more> <more> <rpat>
    <up arrow> <up arrow>
Aux Functions
    <4> <enter>
    Single Trans
    <off>

Analysis mode
    <CONTINUOUS>
Timing mode
    <SYNC>
Gen Clock
    15
    <INTF>

Data
    <2 -1>
Left display category ERROR (BIT ERRS)
Right display category SIGNAL (DELAY)

```

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**6. Verify that the clock signal from Premisys is present by confirming that the sync light is green, not red.**

## Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)

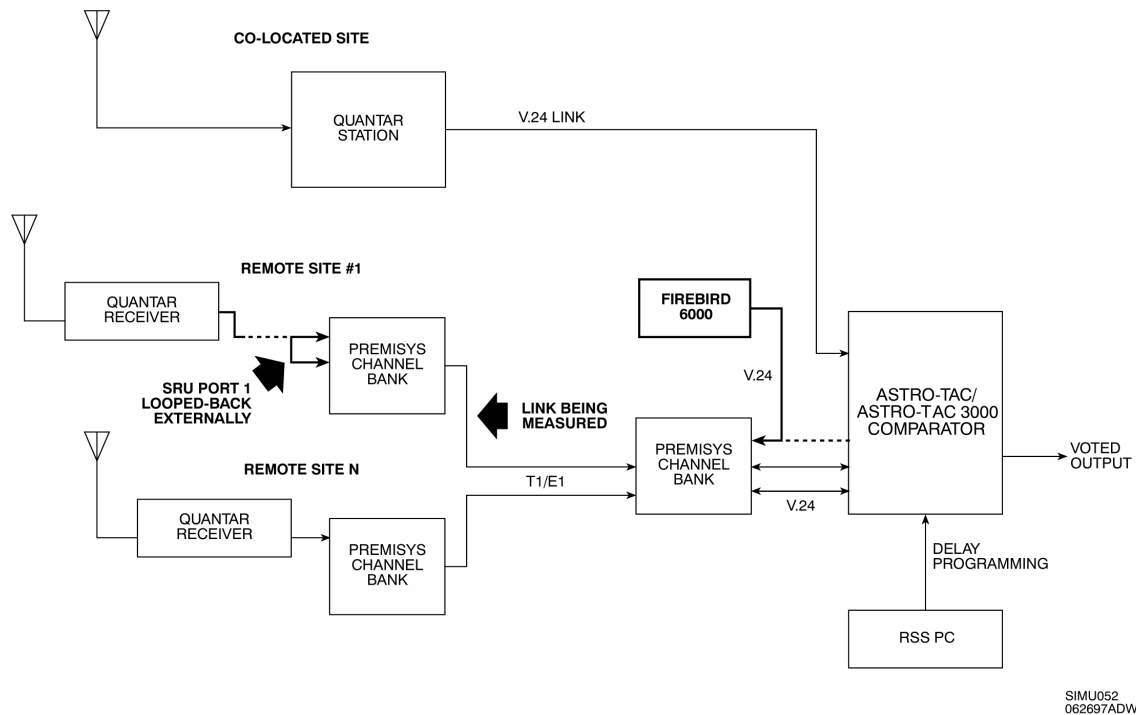


Figure 7-10 V.24 Link Delay Measurement Setup

## Measurement and Optimization (ASTRO-TAC)

1. Press RESTART and read delay and bit errors from the display.
2. Divide the delay value obtained by 2 (to get only single direction). This value is the delay contributed by the Telco infrastructure and equipment. Record this value in site 1 "Telco and Equipment Combined Delay (msec) location in Table 7-12.

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)***Table 7-12 V.24 Delay Value Measurements and Calculations*

Link No.	Telco and Equipment Combined Delay (msec)	Add $\Delta$ (msec)
Site 1	_____	_____
Site 2	_____	_____
Site 3	_____	_____
Site _	_____	_____
Site _	_____	_____
Site _	_____	_____
<b>EXAMPLE:</b>		
Site X	45	—
Site Y	20	<b>+25</b> [greatest delay is 45 msec at site X. Therefore; 45 msec – 20 msec (site Y delay)= delay amount needed to be added to site Y (add $\Delta$ value) to balance site delays]

3. Repeat for other sites.
4. Add delay to the Add  $\Delta$  column so that all links have the same delay as the most delayed site. This results in a zero differential link delay between links.
5. For each link, enter the appropriate link delay compensation (Add  $\Delta$  values in Table 7-12) into the corresponding comparator ports using the comparator RSS.

The field can be found in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG window.

**Measurement and Optimization (ASTRO-TAC 3000)**

1. Press RESTART and read delay and bit errors from the display.
2. Divide the delay value obtained by 2 (to get only single direction). This value is the delay contributed by the Telco infrastructure and equipment. In a copy of Table 7-13, record this value in site 1 "Telco and Equipment Combined Delay (msec) location.
3. Repeat for other sites.
4. For each link, enter the total link delay ("Telco and Equipment Combined Delay column in Table 7-13) into the corresponding comparator ports using the comparator RSS.

The field can be found in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG window.



Table 7-13 V.24 Delay Value Measurements

Link No.	Telco and Equipment Combined Delay (msec)
Site 1	_____
Site 2	_____
Site 3	_____
Site _	_____
Site _	_____
Site _	_____

## Optimize by Analog Link Delay Measurement Using Modems Procedure

Use the following procedure to measure and optimize the delay of an analog link. These facilities typically employ 3002-spec, 4-wire analog leased line and are often used with ASTRO conventional systems equipped with ASTRO wireline modems.

The total link delay on an analog line can be measured using UDS V.3225 analog modems and the procedure described in “Optimize by V.24 Link Delay Measurement Procedure” section located on page 7-61.

### Measurement and Optimization (ASTRO-TAC)

Perform measurement and optimization as follows:

1. Obtain modem delay values as follows:
  - 1.1 See Figure 7-11. At the local site, fabricate a delayless analog link by connecting two V.3225 modems using a short length of 4-wire Telco cable.
  - 1.2 On modem that simulates remote site, connect V.24 tx/rx ports in loop-back configuration.
  - 1.3 Connect the Firebird 6000 to V.24 port on local modem.
  - 1.4 Measure modem delays using Firebird 6000 as described in “Optimize by V.24 Link Delay Measurement Procedure” section located on page 7-61.
  - 1.5 Record the modem delay value determined above in all locations in the Local LB Delay row in Table 7-14.
2. See Figure 7-11. Reconnect local modem to actual line that serves remote site.
3. Configure the local V.3225 modem to generate system clock using the **Internal Transmit** clock selection in the **Modify configuration?:Change modem parameters?:Transmit Clock:Internal** menu.

Set the remote modem to **Transmit Clock:Receive**.

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)**

4. Ensure that the V.24 cables are wired to include the synchronous clocking signal.

**NOTE**

Do not include Telco-provided channel service units (CSUs) in synchronous clocking measurement.

5. (See Figure 7-12.) At remote site, perform wire loopback on V.3225 modem.
6. Using the procedure described in “Optimize by V.24 Link Delay Measurement Procedure” section located on page 7-61, measure the local/remote V.24 link delay (including the CSUs) for the site. Record this reading in the Roundtrip Delay location in Table 7-14 that corresponds to the site.
7. Subtract the back-to-back V.3225 delay measured in step 1 (local LB) from each Roundtrip measurement, divide by 2, and record the resulting value in each site’s Infrastructure Delay location in Table 7-14.  
  
This provides the infrastructure delay contribution of the analog path independent of the modem delays.
8. Depending on the link type, add the appropriate equipment delay value from Table 7-9 (12 msec for ASTRO wireline modems).
9. Calculate the Add  $\Delta$  values so that all links have the same delay as the most delayed link.
10. Enter the comparator delay compensation values (Add  $\Delta$  values from Table 7-14) for each port using RSS.

The field can be found in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG window.

**EXAMPLE:**

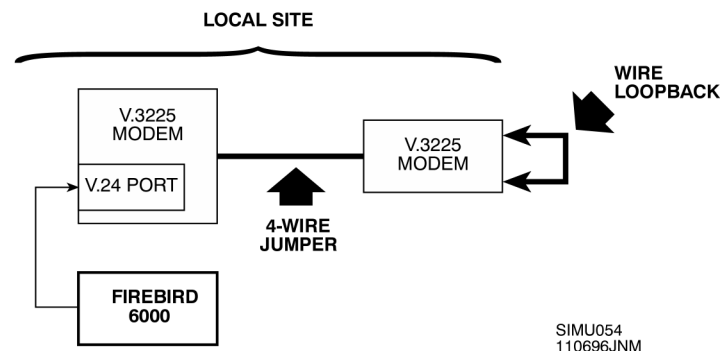
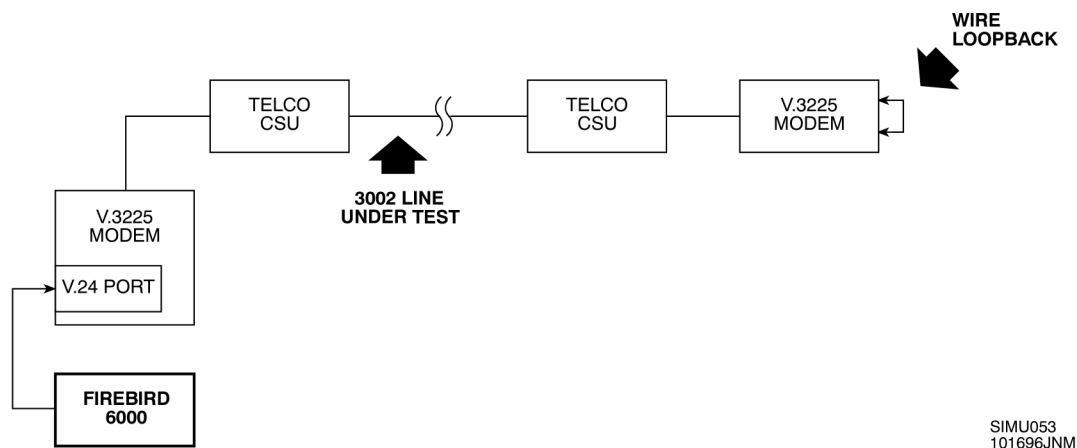
Assume the following values:

- ☐ Measured roundtrip delay =58 msec
- ☐ Loopback (LB) delay =56 msec
- ☐ ASTRO wireline modems =12 msec

Example Values:	Calculations:
$[58 \text{ msec} - 56 \text{ msec}] / 2 = 1 \text{ msec}$	$[\text{Roundtrip Delay} - \text{Local LB Delay}] / 2 = \text{Infrastructure Delay}$
$1 \text{ msec} + 12 \text{ msec} = 13 \text{ msec}$ for example link	Infrastructure Delay + Equipment Delay Value (Table 7-9) = Link Total Delay

Table 7-14 *Delay Calculation Form for Analog Line Measurement*

	Site(Link)No.____	Site(Link)No.____	Site(Link)No.____
<b>1 ROUNDTRIP DELAY</b> (V.24 Link Delay Measurement)			
<b>2 LOCAL LB DELAY</b> (step 1 of this procedure)			
<b>3 INFRASTRUCTURE TOTAL DELAY</b> (subtract line <b>2</b> from line <b>1</b> , then divide by 2.)			
<b>4 EQUIPMENT DELAY</b> (obtain from Table 7-9)			
<b>5 TOTAL LINK DELAY</b> (add lines <b>3</b> and <b>4</b> )			
<b>6 ADD Δ VALUE</b> (noting link with greatest total link delay, obtain add Δ values for other links by subtracting line <b>5</b> of each link from greatest total link delay)			
$[\text{Roundtrip Delay} - \text{Local LB Delay}] / 2 = \text{Infrastructure Delay}$ Infrastructure Delay + Equipment Delay Value (Table 7-9) = Link Total Delay			

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)**Figure 7-11 **Setup For Measurement Of Back-To-Back Modem Delay**Figure 7-12 **3002 Channel Delay Measurement Setup****Measurement and Optimization (ASTRO-TAC 3000)**

Perform measurement and optimization as follows:

1. Obtain modem delay values as follows:
  - 1.1 See Figure 7-11. At the local site, fabricate a delayless analog link by connecting two V.3225 modems using a short length of 4-wire Telco cable.
  - 1.2 For the modem that simulates remote site, connect V.24 tx/rx ports in loop-back configuration.
  - 1.3 Connect the Firebird 6000 to V.24 port on local modem.
  - 1.4 Measure modem delays using Firebird 6000 as described in “Optimize by V.24 Link Delay Measurement Procedure” section located on page 7-61.
  - 1.5 Record the modem delay value determined above in all locations in the LOCAL LB DELAY row in Table 7-14.

2. (See Figure 7-11.) Reconnect local modem to actual line that serves remote site.

3. Configure the local V.3225 modem to generate system clock using the **Internal Transmit** clock selection in the **Modify configuration?:Change modem parameters?:Transmit Clock:Internal** menu.

Set the remote modem to **Transmit Clock:Receive**.

4. Ensure that the V.24 cables are wired to include the synchronous clocking signal.

<b>NOTE</b>
-------------

Do not include Telco-provided channel service units (CSUs) in synchronous clocking measurement.

5. (See Figure 7-12.) At remote site, perform wire loopback on V.3225 modem.
6. Using the procedure described in “V.24 Link Delay Measurement” section above, measure the local/remote V.24 link delay (including the CSUs) for the site. Record this reading in the ROUNDTRIP DELAY location in Table 7-14 that corresponds to the site.
7. Subtract the back-to-back V.3225 delay measured in step 1 (local LB) from each Roundtrip measurement, divide by 2, and record the resulting value in each site’s INFRASTRUCTURE TOTAL DELAY location in Table 7-14.  
  
This provides the infrastructure delay contribution of the analog path independent of the modem delays.
8. Depending on the link type, add the appropriate equipment delay value from Table 7-9. (12 msec for ASTRO wireline modems).
9. Enter the comparator delay values (Total Link Delay values from Table 7-14) for each port using RSS.

The field can be found in the MAIN:VIEW/EDIT CPLG:INFRA PORT CONFIG window.

**Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)****EXAMPLE:**

Assume the following values:

- ☐ Measured roundtrip delay =58 msec
- ☐ Loopback (LB) delay =56 msec
- ☐ ASTRO wireline modems =12 msec

Example Values:	Calculations:
$[58 \text{ msec} - 56 \text{ msec}] / 2 = 1 \text{ msec}$	$[\text{Roundtrip Delay} - \text{Local LB Delay}] / 2 = \text{Infrastructure Delay}$
$1 \text{ msec} + 12 \text{ msec} = 13 \text{ msec}$ for example link	Infrastructure Delay + Equipment Delay Value (Table 7-9) = Link Total Delay

Table 7-15 **Delay Calculation Form for Analog Line Measurement**

	Site(Link)No.____	Site(Link)No.____	Site(Link)No.____
<b>1</b> <b>ROUNDRIP DELAY</b> (V.24 Link Delay Measurement)			
<b>2</b> <b>LOCAL LB DELAY</b> (step 1 of this procedure)			
<b>3</b> <b>INFRASTRUCTURE TOTAL DELAY</b> (subtract line <b>2</b> from line <b>1</b> , then divide by 2.)			
<b>4</b> <b>EQUIPMENT DELAY</b> (obtain from Table 7-9)			
<b>5</b> <b>TOTAL LINK DELAY</b> (add lines <b>3</b> and <b>4</b> )			
$[\text{Roundtrip Delay} - \text{Local LB Delay}] / 2 = \text{Infrastructure Delay}$ Infrastructure Delay + Equipment Delay Value (Table 7-9) = Link Total Delay			

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## Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping)

<b>NOTE</b>
-------------

This section applies only to the ASTRO-TAC 3000 comparator **with** ping feature. For ASTRO-TAC and earlier ASTRO-TAC 3000 comparators, go to “Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)” section.

Ping availability on an ASTRO-TAC 3000 comparator is verified by version V03.01.00 (or later) appearing on RSS window.

### Optimization Procedure

Perform inbound optimization as follows:

1. (Trunking only) Disable the trunked channel that is being aligned on the Prime Central Controller (TIB).
2. Connect the laptop PC serial port to the terminal connector on the ASTRO-TAC 3000 being aligned.
3. Read the code plug of the ASTRO-TAC 3000 using the RSS.
4. After the code plug is completely read into the service terminal, go to the **Main Menu**.
5. From the **Main Menu**, press the <F2> key to access **Service** menu.
6. Press the <F6> key to access the Status Panel.

**Digital System Inbound Optimization (ASTRO-TAC 3000 with Ping)**

7. Press the <F2> key to access Status Panel Display.

Example status panel display is shown in the following figure.

Port#	Alias	State	HDLC Address	Link Delay	Voted Port
1	CO-LOCATED	IDLE	01		
2	SITE 2	IDLE	02		
3	SITE 3	IDLE	03		
4	NONE	DISABLED	3F		
...					
...					
16					

SIM4061  
0731ADW

8. Press the <F3> key for GET LINK DELAYS.

Link delay values are now measured by ping and displayed in msec.

Example of measured delay values of 0 msec, 20 msec, and 19 msec for Co-loc, Remote Site 2 and Remote Site 3, respectively, are shown in the following figure.

Port#	Alias	State	HDLC Address	Link Delay	Voted Port
1	CO-LOCATED	IDLE	01	000	
2	SITE 2	IDLE	02	020	
3	SITE 3	IDLE	03	019	
4	NONE	DISABLED	3F		
...					
...					
16					

SIM4062  
0731ADW



**NOTE**

Base Stations/Receivers must be ping-compatible, in addition to ASTRO-TAC 3000, for ping to function. If the idle link does not return delay value, it can be assumed that the Base Station/Receiver is not ping-compatible. In this case, perform manual link delay inbound optimization in accordance with “Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)” section earlier in this chapter.

As a quick check of link integrity, step 8 can be repeated several times. Repeated GET LINK DELAYS operations should return delay values within 2 msec tolerance for a given link.

9. Press the <F8> key for SAVE LINK DELAYS to save link delays.  
Service terminal displays SAVING LINK DELAYS TO CODEPLUG IMAGE IS COMPLETE.
10. Press the <F10> key to exit current window.
11. Press the <F10> key to exit Status Panel.
12. Press the <F10> key to exit to **Main Menu**.
13. Press the <F3> key to access **Get/Save/Change Codeplug** menu.
14. Press the <F8> key to program the setup into codeplug.  
At the Proceed with programming? prompt, press the <F2> key to proceed with programming.  
At the Reset Device? prompt, press the <F2> key to continue and reset device.
15. Press the <F10> key to exit to **Main Menu**.
16. Exit the RSS program and repeat steps 1 through 15 for any additional ASTRO-TAC 3000 comparators in the system.

## Logging Delay Values in System Log

Whenever the ping feature is used, it is recommended that the delay values displayed be recorded in a system log. In this manner, the integrity of the links can be assessed. A properly functioning infrastructure should not experience significant changes in delay over time (i.e., over 5 msec).

The log should note the following data:

- ☐ Date
- ☐ Link alias and type
- ☐ Delay value for each link

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## Digital (ASTRO) System Outbound Optimization

The following procedure optimizes outbound (station to mobile) simulcast operation in ASTRO systems using the ASTRO-TAC/ASTRO-TAC 3000 comparator.

<b>NOTE</b>
-------------

Properly calibrated equipment must be used throughout the procedures in this section. Where feasible, use the same piece of equipment for each measurement. If not feasible, or if more than one technician must work at the same time, the same model equipment should be issued to all technicians involved in this procedure.

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### Digital Optimization

This section discusses the procedures required for optimization of the ASTRO portion of the simulcast system.

<b>NOTE</b>
-------------

(Trunking only) During optimization, it is recommended that an operator be present at the Zone Manager to watch for alarm conditions (such as loss of GPS, 1 PPS, 5 MHz or ASTRO links) which would interfere with system optimization or cause invalid results.

---

(Trunking only) Prior to any adjustment or optimization at a site, the channel being optimized must first be removed from service by the 6809 or Zone Manager.

---

**Digital (ASTRO) System Outbound Optimization****Station Coarse Deviation Set**

It is important that the station deviation be properly aligned. Follow the deviation alignment procedure found in the Quantar **F2 Service** menu.

This procedure involves transmitting on one frequency at a time on a group of frequencies. Then, the deviation (as measured on a service monitor or mod analyzer) are entered into the RSS computer.

The station digital-to-analog converter produces a constant output tone for this measurement. This tone is routed by the station to the modulator. The deviation level of this tone is measured by a service monitor or mod analyzer. Doing this allows the station to understand the modulation sensitivity characteristics of the Voltage-Controlled Oscillator (VCO). The modulation sensitivity of the VCO varies with frequency.

This is the only deviation set required for simulcast in the ASTRO digital mode.

**Site GPS Operational**

Verify that the GPS receivers at the prime and remote sites are installed and operational. This may be accomplished in any one of several ways as follows:

- ☐ Use Zone Manager (or prime site SMT) current alarms may be used to determine that all site GPS standards are properly operating.
- ☐ Examine the GPS receivers front panel indicators at each of the sites.
- ☐ Use the serial port of the site GPS receivers to determine that the GPS receiver is properly operating and has satellite lock.
- ☐ Examine the external output alarm reporting from the GPS standards.

**Station and Comparator 1 PPS and 5 MHz Operational**

Verify that the stations all have operational 1 PPS and 5 MHz signals. Verify that the comparators have operational 1 PPS. (5 MHz is not needed by comparators). This can be accomplished in either of two ways as follows:

- ☐ Use Zone Manager (or prime site SMT) current alarms to determine that all site GPS standards are properly operating.
- ☐ Examine the RSS error log can at each station and comparator.

**Comparator Launchtime Parameter**

This procedure adjusts the fixed delay value that the comparator adds to the current time prior to time stamping data that is sent to the station.

1. Connect to the prime site ASTRO-TAC/ASTRO-TAC 3000 with ASTRO-TAC/ASTRO-TAC 3000 RSS and enter the **CHANGE/SYSTEM FEATURES/SIMULCAST PARAMETERS** menu.
2. Enable the simulcast function.
3. Change the signal launchtime offset parameter to the value given in Table 7-16, depending on link type.

Table 7-16 *Launchtime Offset Values Based On Link Type*

Link Type	Launchtime Offset	
	Voice	Data
Premisys (NOTE)	165 msec	220 msec
Siemens	126 msec	181 msec
V.3225 modems	148 msec	203 msec
ASTRO wireline modems	132 msec	187 msec
<b>NOTE:</b> For Premisys with V.2.6 firmware, Voice and Data values are 145 msec and 200 msec, respectively.		

The values in Table 7-16 are not determined solely by the link and box throughput delays. These values are determined empirically to be a value that also allows adequate station buffers for proper operation with console takeover operations. Improper setting of this variable could ultimately result in station de-keys and simulcast errors when station buffers grow too large (and the data too old) after multiple console takeovers.

**NOTE**

The procedure in the “Optimize by V.24 Link Delay Measurement Procedure” section located on page 7-61 can be used to measure and determine the offset adjustment for a high delay infrastructure (such as systems that use SONET). This procedure is not normally required because the delay contributions of the telephone company or microwave transport infrastructure are normally very small compared to those added by the Motorola equipment.

**Digital (ASTRO) System Outbound Optimization****Detail of Link Delays Affecting Launchtime**

The total link delay on the synchronous V.24 links is the result of the combination of a number of contributing delay factors. Table 7-17 lists the typical delay factors.

*Table 7-17 Typical One-Way Delays For Various Link Types*

Link Type	Typical End-to-End Delay (One-Way)
Telco Controlled:	
Copper T1	8-10 $\mu$ s/mile
Fiber Optic T1	10 $\mu$ s/mile
Microwave T1	7 $\mu$ s/mile
CO DAX	125-500 $\mu$ s/switch
Customer Controlled:	
Premisys SRU	45 msec/link (NOTE 1)
SRU+ADPCM (V.24 path)	37 msec/link (NOTE 2)
Siemens DSU sync 9.6	3-6 msec/link
<b>NOTE:</b> 1. If SRU contains V2.6 firmware, delay is 22.5 msec/link instead of 45 msec/link. More accurate value is obtained, where available, using ASTRO-TAC 3000 with Ping. 2. If SRU contains V2.6 firmware, delay is 15 msec/link instead of 37 msec/link.	

As an example, assume a worst case link delay budget for the Telco-controlled portion of the simulcast subsystem as follows:

Link Type	Distance or No. Units	Accrued Delay
Copper T1	100 mi.	1.0 msec
CO DAX	7 switch units	4.0 msec
<b>TOTAL LINK DELAY:</b>		<b>5.0 msec</b>

For the customer-controlled portion of the same example subsystem, assume the following delays:

Infrastructure Equipment	Delay	
	Voice	Data
ASTRO-TAC/ASTRO-TAC 3000 and station tx	95 msec	150 msec
Premisys SRU	45 msec	45 msec
Station buffer allowance	20 msec	20 msec
<b>TOTAL DELAY:</b>	<b>160 msec</b>	<b>215 msec</b>
<b>NOTE:</b> If SRU contains V2.6 firmware, delay is 22.5 msec/link instead of 45 msec/link.		

From the calculations in the proceeding table, it can be seen that the telephone infrastructure delay contribution (5 msec) to the system delay is so minor that it can often be approximated as simply a 5 msec delay contribution. If a 5 msec delay is anticipated due to unusual infrastructure (such as SONET), the procedure described in the “Measurement of V.24 Link Delays” section that follows can be used to measure the delay and adjust the comparator launch time delay outward accordingly.

### Outbound Measurement of V.24 Link Delays (Optional)

The total link delay on the synchronous V.24 links can be measured if needed. This is usually only necessary in unusual cases, such as when packet switched voice or SONET infrastructure are used and the delay is very high compared to that of the Motorola equipment.

### Equipment Required

PN/Description	Manufacturer
Firebird 6000 Telecommunications Test Set	Telecommunications Techniques Corp.

### Setup

1. Disconnect the ASTRO-TAC/ASTRO-TAC 3000 from the prime site channel bank. Connect a channel bank V.24 port to the Firebird rear panel V.24 connector.
2. Disconnect the corresponding V.24 port at the remote site channel bank from the connected station. Loopback the tx and rx data pins on the V.24 cable connected to the appropriate channel bank data port.

**Digital (ASTRO) System Outbound Optimization****3. Turn on the Firebird and select:**


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```

      <Menu>
          Intf Setup
          <INT232>
          <EMLATE>
          <DTE>
          Test Interval
          <10E4>
          Char Format
          (not required; just press <enter>)
      Auxiliary (Aux Functions)
          <3> <1> <enter>
          Delay
          <start> <more> <more>
          <gpatt> <up arrow> <stop>
          <more> <more> <rpat>
          <up arrow> <up arrow>
      Aux Functions
          <4> <enter>
          Single Trans
          <off>

      Analysis mode
          <CONTINUOUS>
      Timing mode
          <SYNC>
      Gen Clock
          15
          <INTF>

      Data
          <2 -1>
      Left display category ERROR (BIT ERRS)
      Right display category SIGNAL (DELAY)
  
```

---

**4. Verify that the clock signal from Premisys is present by confirming that the sync light is green, not red.**



## Measurement and Optimization

1. Press RESTART and read delay and bit errors from the display.
2. Subtract the channelbank contribution to the V.24 round-trip delay ( $2 \times 45$  msec = 90 msec for Premisys) from the delay value.  
  
Divide this delay value by 2 (to get only single direction). This value is the Telco infrastructure, provided one-way delay contribution.
3. Add the Telco delay contribution to the value listed in Table 7-9 for the link type used. Record this value.
4. Connect to the prime site ASTRO-TAC/ASTRO-TAC 3000 with ASTRO-TAC/ASTRO-TAC 3000 RSS and enter the **CHANGE/SYSTEM FEATURES/SIMULCAST PARAMETERS** menu.
5. Enable the simulcast function.
6. Change the signal launchtime offset parameter to the value determined in measurement step 3.

## Moving the Zero Phase Error Contour (Optional)

### NOTE

This step should not be attempted without first consulting a Motorola area systems engineer (ASE) who can provide the proper parameters, if required.

Launch time offset is provided to allow a station to transmit at a time offset from the comparator-generated launch time. This parameter can also be used to compensate for differing transmitter antenna cable lengths and RF combining delays. Adjust launch time offset value as follows:

1. Connect to the sub-site Quantar station with Quantar RSS and enter the **MAIN/SERVICE/ALIGNMENT** menu.
2. Enter the ASE supplied parameter into the ASTRO Simulcast Launchtime Offset field.
3. Repeat this procedure for each station at the affected sites as specified by the ASE.

The range of delay is zero to 975.2  $\mu$ s in 100 ns steps. The time is entered in the RSS in units of tenths of microseconds.

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## Mixed System Optimization

Mixed systems use DSM-II modules for analog channels and ASTRO-TAC/ASTRO-TAC 3000 comparators for ASTRO channels. Analog and digital channels are individually optimized in accordance with the respective “Analog System Optimization” and “Digital System Inbound Optimization (ASTRO-TAC/ASTRO-TAC 3000)” sections of this chapter.

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## Performance Verification

Soon after optimization, performance can be verified by both watching system alarms and by conducting on-the-air testing as described in the following sections.

### Infrastructure Alarms

The Zone Manager, comparators and stations should be examined for alarms. Most products contain alarm logs that can be useful in diagnosing system problems.

If the Quantar base station is not able to launch audio at the time specified by the ASTRO-TAC/ASTRO-TAC 3000 or comparator (i.e., station told to launch after time has passed), an alarm is generated. This alarm can be seen at the Zone Manager as well as in the local station RSS. If this alarm occurs, the signal launchtime offset parameter in the comparator needs to be increased. If the cause for these alarms is not readily apparent, perform applicable optimization or link delay measurement procedure.

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## Optimization Using A Remote Console

In many simulcast systems, the optimization console cannot receive the test signals from all the remote site transmitters. In cases where moving the optimization console to a remote site still does not solve this problem, multiple remote consoles are required throughout the system. When using multiple consoles, at least one transmitter site must be common to both console coverage areas.

The procedure for optimization using a remote console is described in “Trunked Radio System Dual Path and Digital Path Simulcast Equipment” (Motorola manual 68P81081E60). When using the procedure, note the following:

- ☐ Where procedure discusses “wide band modem,” this refers to the DSM-II module.
- ☐ When aligning the AOU, leave the TLPs in the DSM-II at 0 dB.
- ☐ Omit the “Phase Optimization” procedure.

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**Optimization Using A Remote Console**

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